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# MUSIC PRODUCER'S GUIDE TO MIXING

Digital Edition

FUTURE

FIRST EDITION

## MASSIVE GUIDES

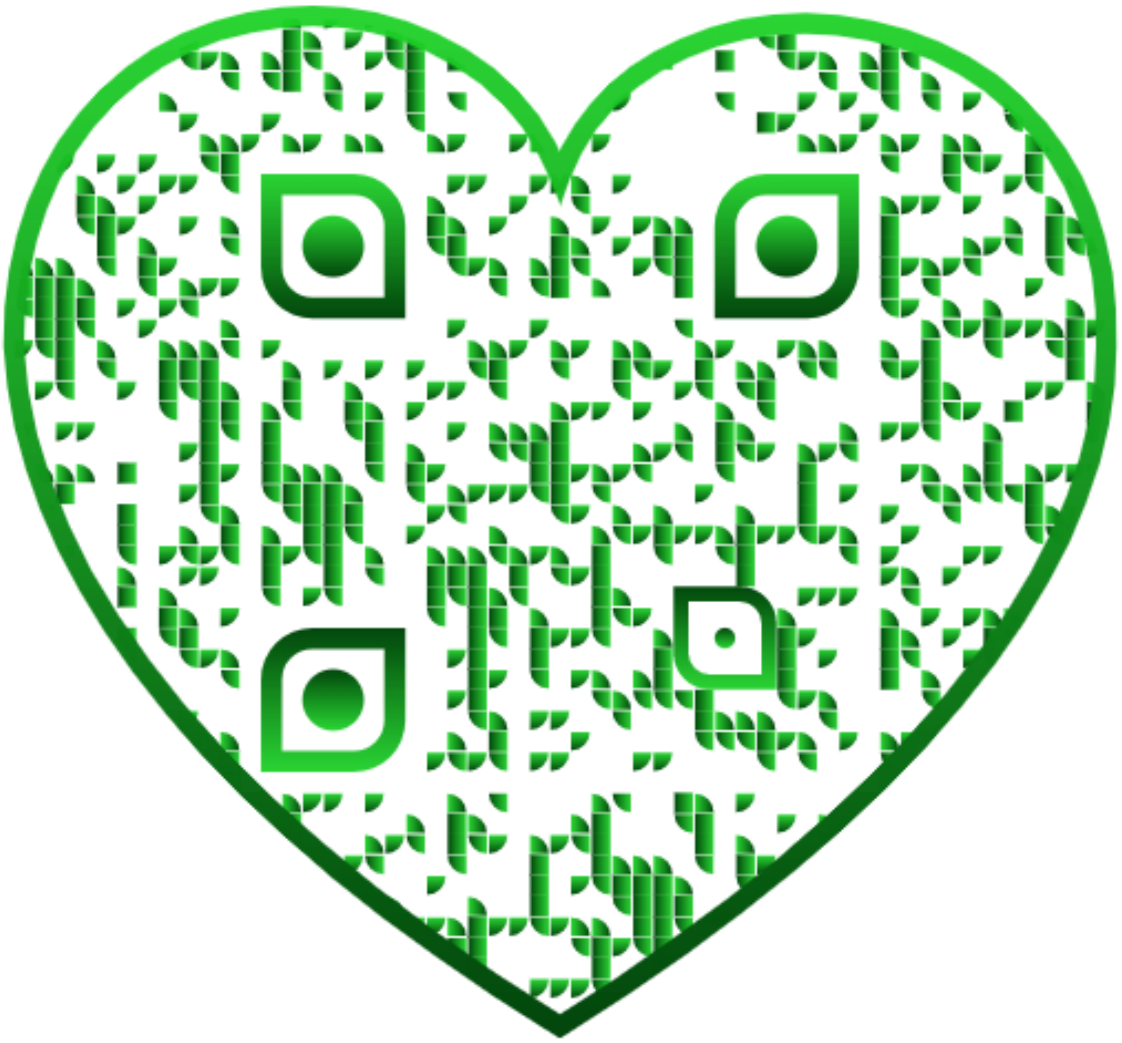
STEP-BY-STEP TUTORIALS FOR A PRO SOUND

## 80 FREE PLUGINS

TO CREATE THE PERFECT MIX

## IN DEPTH TUTORIALS

AUTOMATION • DYNAMICS • VOCAL MIXING • ANALOGUE FEEL



# MUSIC PRODUCER'S GUIDE TO MIXING

Welcome to the ***Music Producer's Guide to Mixing***! This is the complete guide to helping you get perfect, professional-sounding tracks, no matter what music you make, and is brought to you by the experts at *Computer Music* magazine.

It used to be the case that you could only really get a professional-sounding mix after years of experience working in a 'proper' recording studio with a (often huge) hardware mixing desk. Nowadays you get all of that mixing power – plus a lot more besides – in your software DAW (Digital Audio Workstation). So if you use one of the many DAWs out there – Cubase, Logic, Ableton Live, FL Studio, Bitwig Studio. Tracktion Waveform, or one of many more – then this guide is for you, and you already have everything you need to create perfect mixes on your PC or Mac.

All you need in addition to this software are a few mixing ground rules and insider tips and tricks, and we're bringing you all of these in this special *Music Producer's Guide To Mixing* bookazine. We guide you through the basics on p10 with our Fast Guide To Mixing, and then take you step-by-step through a complete professional mix in our Fix Your Mix feature on p12. There's everything you need right there to create a pro-sounding mix, and we even give you all of the plugins mentioned in the feature in a **huge, free 80-plugin download** that comes with this guide. It's the world-famous *Computer Music* Plugin Suite and has all of the mix effects you need, plus lots of synths, drum machines and samplers.

Elsewhere we guide you through some other mixing essentials, including automation (p52), dynamic processing (p92), and vocal mixing (p58). There are masterclasses on several detailed mixing themes, such as adding width to your mixes, mixing bass and kick drums and using reverb on vocals – all essential techniques to learn for great-sounding mixes.

We close with an extra-special guide on how to bring an analogue and more human sound to your mixes. Using a computer can lead us down a too digital path, and this feature will help you bring warmth and character to your music.

As well as the 80+ plugins in the *Computer Music* Plugin Suite, we are also giving you **over 3,000 samples** with which you can create complete songs and practice the new-found mixing techniques that you will hopefully – definitely! – pick up from this guide.

Enjoy this special mixing issue and I hope many of the features help to bring a professional sheen to your music. Keep on mixing!

Andy Jones, editor

FUTURE

# MUSIC PRODUCER'S GUIDE TO MIXING

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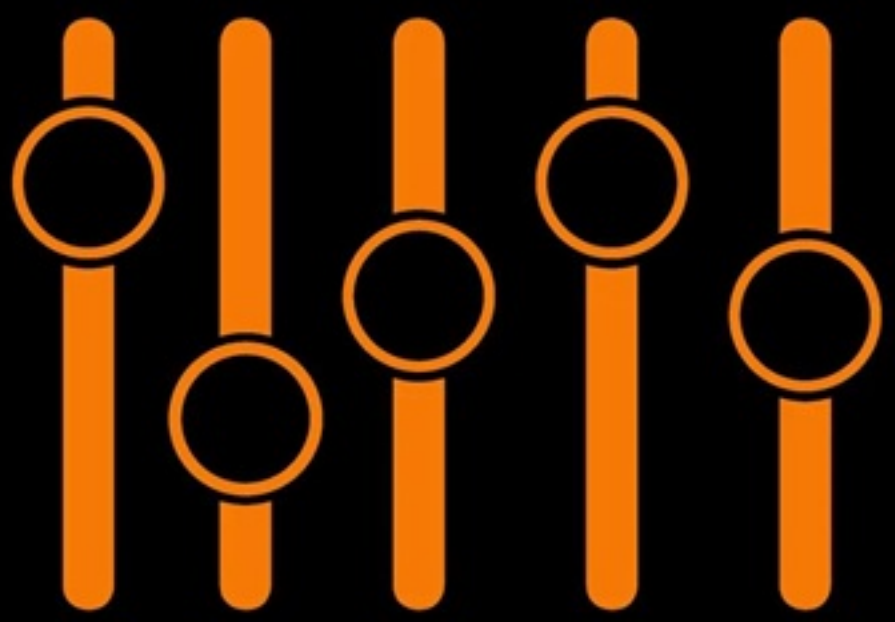
# MUSIC PRODUCER'S GUIDE TO MIXING

Hands-on masterclasses, in-depth features and quick-fire tips – everything you need to mix great music... now!



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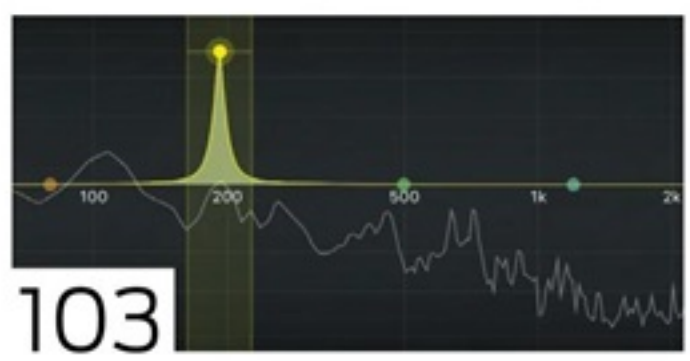


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# How To Use This Guide

Here's your step-by-step guide on how to make the most of this special mixing bookazine. We detail all of the videos, software and tutorial files that accompany each of the features, all delivering a completely interactive music mixing experience!

**> Many of the features in this Music Producer's Guide To Mixing are accompanied by audio examples and video files to help you work through them. All of these videos and audio files are available to download at www.filesilo.co.uk. Audio or sound files will indicate how certain production techniques will impact on the tune you are making or the sound you are creating, and the videos will guide you through many of the workshops in case you get stuck.**

**We have also included the world-famous *Computer Music* Plugin Suite, a set of over 80 instruments and effects that you can download and install into your DAW (either the Mac or PC version) completely free of charge. Many of these are used in the workshops in this guide – like the Fix Your Mix tutorial from p12.**

In order to access all of this content, go to filesilo.co.uk and select the Bookazines tab and find this issue. You will be presented with a main Tutorial Audio folder (for the audio example files) and Video Tutorials folder (for the example videos). The *Computer Music* Plugin Suite of instruments and effects is in the Software section and comprises instruments and effects for Mac or PC. Simply download the plugins for whichever type of computer you use. You will also be able to download your free samples in the Miscellaneous section – see opposite.

More detailed instructions on how to install the software instruments and effects are included on the opposite page. Enjoy!

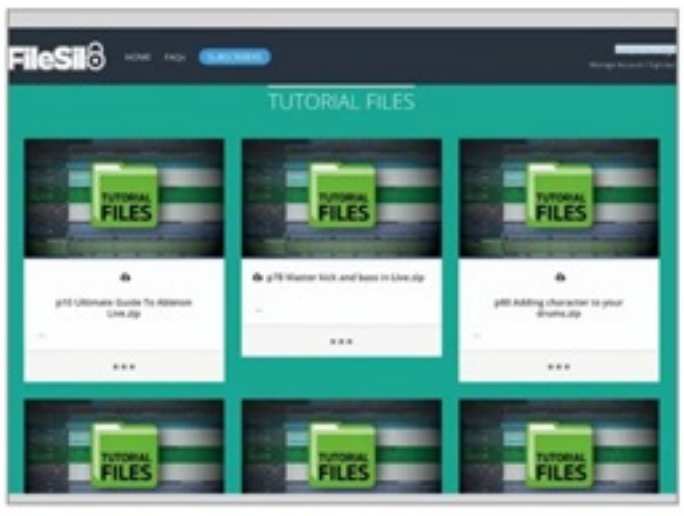
## >Step by step Using FileSilo

1 > Head over to www.filesilo.co.uk. Here are a number of magazine brands and all the digital content associated with them for you to download.

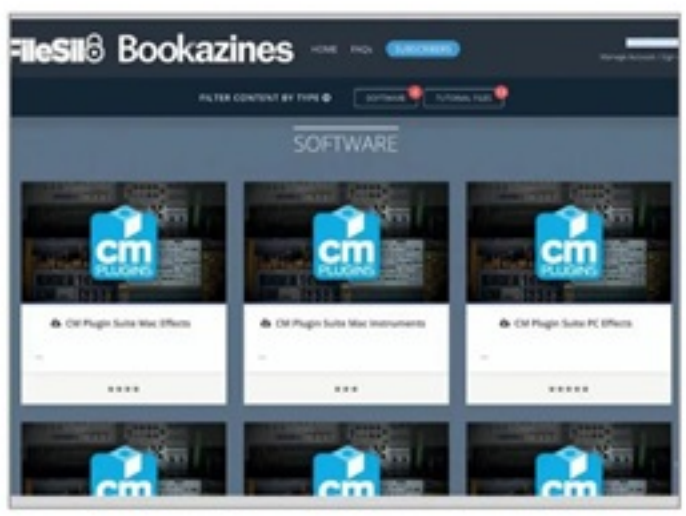
2 > Go to the Bookazines tab at the main filesilo.co.uk screen. Click this tab and you will see a list of recent Bookazines we have produced including The Music Producer's Guide To Mixing. Click on this – it will have the same image as the cover of this bookazine.

3 > Once you are in, scroll to the bottom to the 'Miscellaneous' section. Here you will find your free samples to download, including our Ultimate Vocals pack. You can use these samples in your own songs and apply some of our mixing techniques as you use them.

4 > In the 'Video Tutorials' section we have videos to go with three masterclasses (Stereo Width on page 28, EQ-ing on p48 and Mixing Reverb on Vocals on p78) plus our Get Dynamic feature on p92.




5 > In the 'Tutorial Files' section you will find example audio for many of other features in one folder. These files are designed to show you how some of the mixing processes that we discuss affect your music productions.




6 > Scroll down and in the main 'Software' section you will find all of the plugins in the *Computer Music* Plugin Suite. These are listed as Mac or PC instruments and effects. There's also a guide to the suite as a downloadable PDF. You can find out more information about installing them on the opposite page.

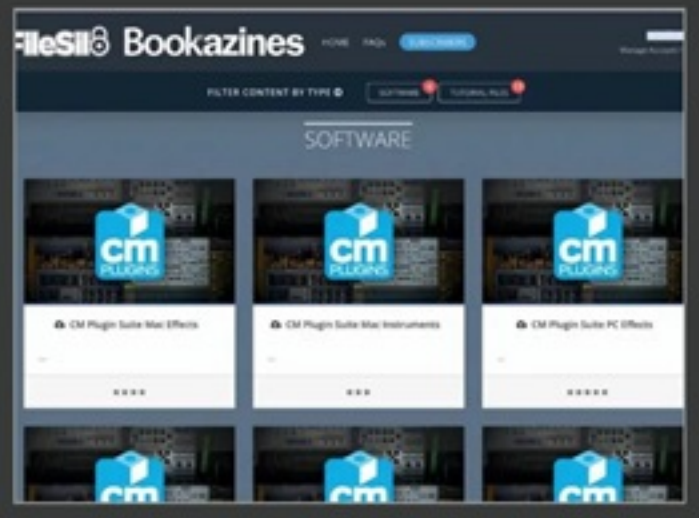
# Your free samples

We have over 3000 samples for you to download and practise your mixing skills with. Here's a quick run down of what you can expect...

**We have over 3000 free audio samples for you to download at www.filesilo.co.uk/Bookazines. Simply select the *Music Producer's Guide To Mixing* and scroll down to the Miscellaneous section which is usually at the very bottom. Here you'll find a couple of synth packs, an Ultimate Vocals pack, nine Track builder packs (with complete song parts to mix and match) plus some bass, drum machine and guitar packs. In short, you get everything you need to create entire pieces of music – beats, bass, guitar, synth and vocals! All the samples were produced by our sample maestros Groove Criminals and Cyclick Samples. They can be used in whatever music productions you like and are copyright free. All we ask is that you don't redistribute the sample packs.**

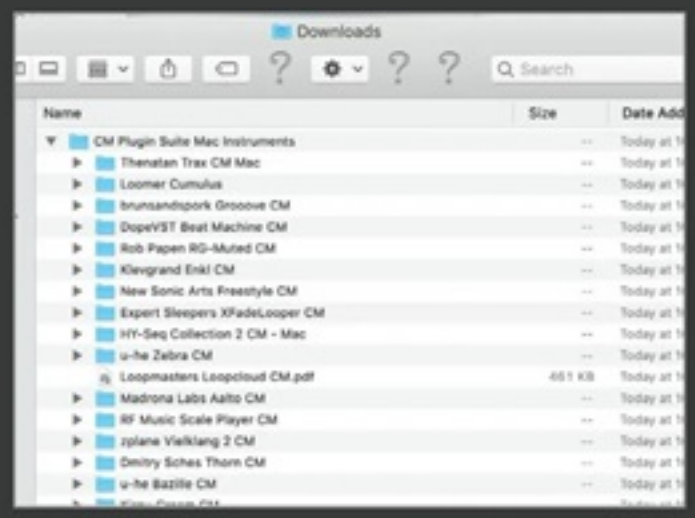
# Installing your free plugins

How to access and install the 80+ free plugin instruments and effects from the *Computer Music* Plugin Suite!

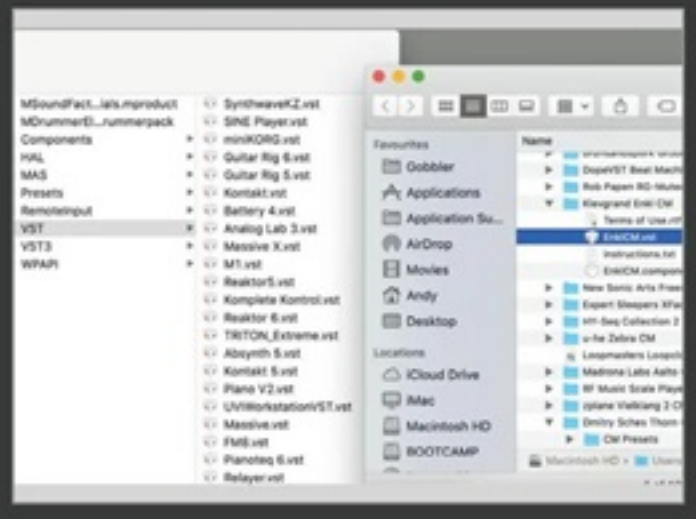



1 > Go to filesilo.co.uk and find this issue under the Bookazines tab. You can find the Plugin Suite under the Software section. There are four main sections for Mac and PC instruments and effects.

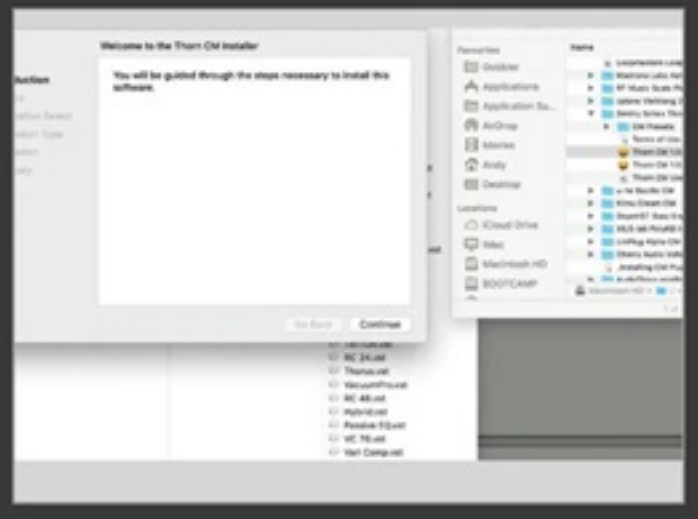
2 > Now simply download whichever instruments and effects you want in blocks. These are big downloads – 1 to 3GB – but you'll get the entire suite in just two downloads (either Mac or PC, depending on your computer).




3 > The complete folder of instruments or effects will download as shown above (on a Mac). Open it up and select which plugin you want to install.




4 > There are generally two ways to install a plugin. If it is shown as a .vst file it is ready to simply place in your VST folder and should work straight away once you start your DAW.




5 > Some, like Thorn, need installing so you need to double click on the file and run through your standard installation procedure. The plugin should install into the correct location.

6 > Now open up your DAW and the installed plugins should appear in the plugins folder. Simply drag whichever one you want onto a MIDI track and you are ready to go!

# The Fast Guide To Mixing

So what exactly is mixing and why is it so important in music production? Find out in our super-fast guide to the perfect mix!

**> This *Music Producer's Guide To Mixing* is full of features on mixing songs: the techniques, tips, processes and tools to create a professional-sounding mix. But before we dive right in, let's go back to basics and examine what mixing is all about. Even if you are a seasoned producer, sometimes going back to the first principles of a subject can be a good idea; maybe we lose site of why we are doing something, having spent years doing it! And if you are new to mixing, you have come to the right place – this feature looks at what mixing is all about: why you mix, how you mix and what you are aiming for with a great mix.**

## Mix it up

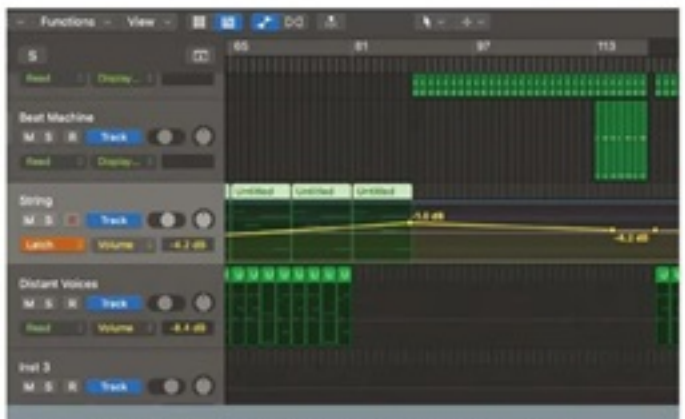
Whenever you create a song in your DAW (or Digital Audio Workstation – think Cubase, Logic, Ableton Live and so on), then you will need to know about the basics of mixing. As soon as you have more than one instrument (or track) in your song, you will need to mix them together to sound coherent and so one doesn't adversely affect the other. With multiple tracks – bass, lead, vocal, guitars, drum and so on – mixing gets more complex, as you have to try and balance all of the sounds so that they sound good and aren't fighting one another.

So mixing starts as soon as you get more than one track in a song. The most obvious way to start mixing is by adjusting the volume of each instrument or vocal part of your multitrack song. In the old days, these tracks were fed through (often large) hardware mixing desks, the volume of each channel being controlled by the mixer fader.

In your DAW, this process is mimicked with a virtual mixing desk, with faders to control each channel's volume. DAWs have audio and MIDI tracks, with the former carrying digital audio by way of recordings from the outside world or samples loaded into your song. The MIDI tracks trigger internal software instruments (synths, samplers etc) or external hardware MIDI instruments, the audio of which is fed back in and recorded as audio. The mixer faders control the volume levels of each type of track. Even with this simple theory, you can achieve a decent mix by adjusting channel volumes, with nothing dominating the song and everything nicely balanced. But that is just the start...

## Your mix in several dimensions

You can think of your stereo song mix in two dimensions at first: up, down, left and right. The up and down refer to the volumes of each channel and the left and right the pan positions. You can easily push all of your song tracks more left or right using each mixer channel's pan control. This is a great way to achieve separation of similar sounding tracks. Say you have two lead guitar tracks that sound very similar, or two backing vocal tracks. Simple nudge one of each left and the other right, and you will hear each



Mix automation is great for altering parameters automatically. Here we're drawing in how our volume increases and decreases on an instrument channel

### Mix automation

**In the old days of hardware mixing, engineers had to record fader and other mixer movements by hand – volume changes for example. Nowadays, in your DAW you can record these movements in real time so that they play back automatically when the mix replays. Better still you can just draw in lines to control parameters (see above). This is known as mix automation and it doesn't just apply to volume and pan! Anything that can be controlled, from effect parameters to knobs on an instrument, can be recorded or automated, so you can tweak intricate parts of a mix in as much detail as you like.**

KRK_s GoAux speakers deliver a great sound at a good price – just one of many monitor options for mixing

## The importance of monitors when mixing

**When mixing, it's important to listen to the end results through a decent set of monitors or headphones. And when we say 'decent', we really mean 'accurate'. What you hear must be an honest reflection of what you are mixing. Cheaper monitors often colour parts of the frequency range – the bass, for example – so that they sound better. This is not good for mixing. If you hear an enhanced bass when mixing, you will pull back on the bass, resulting in bass-thin sounding mixes on any other system you play it on. Your mix monitors should deliver an uncoloured, flat response so you can hear the truth!**

**Fortunately, flat response monitors don't cost as much as they used to and you can get great monitors for mixing at a decent price (£200 upwards) from companies including JBL, Adam Audio, IK Multimedia, KRK, Eve, Presonus, Kali Audio and many more.**

more clearly within the resulting stereo mix, as each is panned away from the other.

The idea of panning, then, is to spread your song out, track by track, so you can hear them all across the stereo field. A good idea when mixing a song recorded by a band is to imagine them playing on stage. The drummer would be in the middle so you'd keep the drums fairly central as you would the main vocal, with guitars spread left and right. There's no hard and fast rule with mixing though – although lower frequency instruments like basses tend to stay central. The main rule is that it's great to hear every track do its thing in a song, and a good mix will allow every instrument or vocal to shine. Oh, and if it sounds good, it IS good!

## EQ time

If you think of panning and volume as two mix dimensions, you can add a third dimension care of an EQ, which adjusts the frequency of each channel. Having an EQ plugin on each channel allows you to cut or boost different frequencies of an instrument or vocal – on a very simple level, the bass and treble – so allowing tracks to stand out by frequency, so you don't always have to resort to adjusting the volume.

This is a great way, for example, of making two low frequency sounds coexist in a mix, even if they are panned centrally. It's often the case that the kick drum and bass guitar sit in the middle of a mix with both occupying lower frequencies. You simply tweak one up in frequency range using an EQ and the other down, and then both become more distinct and stop clashing with one another.

Another very commonly used effect is the compressor. The compressor attenuates (reduces) the volume of a signal once it exceeds a certain level, so is a great way to keep a recording fairly level. You might have a bass guitar recording which really varies in level as it is being played. A compressor helps keep the volume more consistent, making it easier to mix in with the rest of the song.

## Creative effects

The other main effects, aside from compression and EQ, tend to add more creative textures to your sound rather than the more workman-like frequency and volume tamers that are EQs and compressors. Reverbs, for example, add wonderful ambience from rooms or spaces – real or imaginary – delivering the sound of an environment to a recording to make it sound like it was recorded in a famous recording studio or cathedral. Reverbs can add an amazing touch to vocals, guitars and synths.

Delays do exactly what they say on the tin and extend the reflective qualities of reverbs into more distinguishable echoes – great for guitars, basslines and percussion.

There are many other creative effects that can add chorus-like swirls, dramatic filtered frequency effects, low bit-rate digital distortion, analogue saturation (which adds the more gentle distorted sound of old gear to a digital recording), pitch effects to change the tuning of a recording, and even racks of virtual guitar effects which mimic real-world pedals and amplifiers. In short, if you can think of an audio effect, someone will have come up with a plugin to do it in your DAW, so you can apply that effect to any instrument in your mix.

## Less is more

When mixing, one of the key lessons is not to apply too much of anything: volume, effects, panning or any of the principles we have discussed. You start by being subtle, pulling back the volume of every track in your mix and gently adding volume on this track, effects on the other, EQ on another. Take your time, save different versions, take plenty of breaks so you can listen to your music and mixes with fresh ears. Once you are happy, it's time to mix down and extract a stereo file – the finished song. Play this mix on other systems to check that it sounds good everywhere and then, we dare you, play it to other human beings to see what they think!

Mixing can be a very enjoyable process and as creative as you want it to be. Start by making your music sound as great as you can across those two dimensions, explore effects and the depth of the mix and then get as creative as you want with other effects.

And remember, the only real mixing rule is: if it sounds good, it *is* good...

> mix music now / fix your mix

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# FIX YOUR MIX

## Your hands-on guide to producing a complete mix of a pro song! And all the plugins you need to do it are free to download right now!

**> Mixing is traditionally the role of some elusive deity-like genius. A super-hero with supersonic powers. If they can spare the time, and if you can afford them, they appear with a thunderclap before you in a small crater, and blinded by the ethereal light of their plasma sphere you fearfully take a step towards them to hand over The Files...**

You wake, as if from a dream several days later to the ping of your email and, trembling with anticipation, you download the file, take it to your 'system' and hit play. It sounds *amazing*. The *bass*! The space! The drumsound! The vocal! And it's the same in the car. And your earbuds. And your mate's Bluetooth speaker. Perfection!

In the absence of Spike Stent or Andy Wallace though, you'll have to do this all by yourself. And maybe you're not very confident, as your mixes never seem to sound good other than on your own speakers, and even then they sound sh*** compared to *How Deep Is Your Love* by Calvin Harris. Why is that?

It's true that some of the best sounding mixes are made in studios with big analogue consoles and outboard to die for, but it's also true that some of the best bangers have been squeezed out of a laptop on a tour-bus. In reality most good mixes are achieved somewhere between these two extremes and are usually the result of following a few simple rules.

Whether you're fully 'in the box', hybrid or analogue, these rules are universal, and together with a bit of attention to your working environment give you a good shot at creating a solid mix that will translate to the outside world and hold its own against the tunes you love.

This feature will take us through the entire mixing process from multitrack to final mix.

We're using a gentle tech house remix of *Morning Light* by Western Disco as our volunteer session, starting with the raw files, using Logic Pro X as our DAW, and the fabulous and *free* plugins from our own **cm** collection.

We'll talk about what we're doing, and why, so you can translate the techniques in question into your own world, and rest assured, we're staying fully 'in the box', so no £2,000 analogue mix bus compressors or prohibitively expensive outboard DSP plugins here.

Let's dive in!

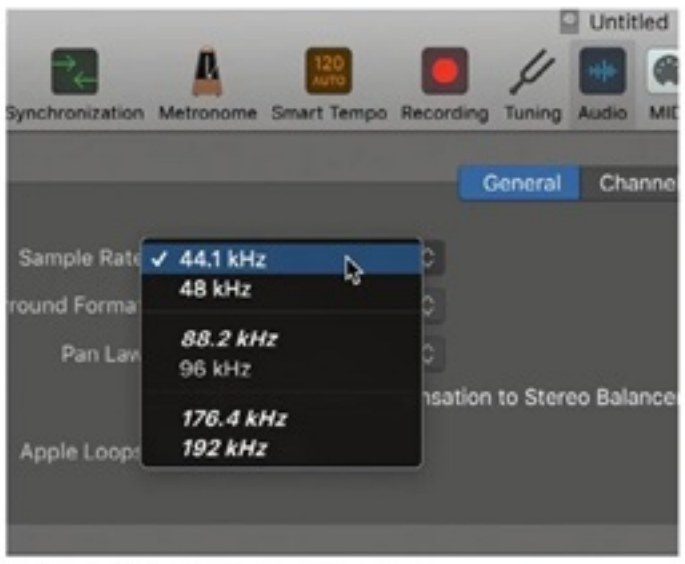
# Sort the multitrack

Let's assume you have your room set up as best you can (see Your Mix Room box). Now you need to get the session mix ready. There's a great argument for flattening your session before you start. If you have a ton of instruments and plugins already eating up your CPU, RAM and USB or Thunderbolt bandwidth playing back complex sampler instruments then why not commit to these sounds as they are and move on.

You're about to mix the track, you're committing then right? So bounce the tracks in place, or render them or do whatever your DAW does to render each track as an audio file. Your computer will thank you for it when you start piling on mix plugins. You can always come back to the pre-mix session and grab whatever you need. So go on... commit!

If you really want to get 'Pro' about it you could bounce a set of wet files (with all the plugs), and a set of dry ones (without them). You can solo and bounce tracks one at a time if they're using shared effects on aux sends, eg a reverb, or you can preserve the same effect send and return setup you were using up till now in your new mix session. The dry files can live on another playlist for easy access in case you unearth a nasty edit or bum note that's gone under the radar. Check your sample rates! If you've been working in 44.1 kHz don't be afraid to up-sample your bounces to 48kHz for the mix, especially if you do happen at all to be using a hybrid setup that includes some outboard equipment.

Start with the wet files as they're going to give you a starting sound that's closer to the rough mix. Bring the stereo rough mix file into your session and line it up with the multitrack. If possible, send the rough mix out of a second pair of outputs into a second return to your monitor controller if you have one. It's a great idea to be able to A/B between your mix and the rough mix so you can feel great about how awesome your mix sounds compared to the rough – or to realise that the rough has way more vibe and you need to start again. If you don't have a hardware option for auditioning a second mix have a serious look at ADPTR's Metric AB plugin, which offers multiple analysis and comparison options for critical listening.




**Sample rate selection in Logic Pro X**

Once you have the new files in place, all at the correct sample rate, same content as the rough mix, then you're good to go. Save the session as 'Mix setup – files in', then save again as 'Mix setup – check multi'. That way you can always go back if you get into a pickle. It happens. Move all the faders except the master down from 0dB to -10dB. This will ensure that, when you hit play, the tracks won't overload the output bus. Spend a minute making a making sure that everything is at a ball park level and you can hear everything you are supposed to.

## Clicks, pops, edits, phase, timing, tuning, noise.... fix it!

Now is a good time to check for unwelcome stuff. Bad edits can cause nasty clicks. Most of the time these can be fixed by making a copy of the small section of audio and redrawing the waveform in the audio editor to smooth out the spike or vertical cut. These little high frequency nasties will only become more apparent as you apply EQ and compression while building the mix, so try and get rid of them early on.

If there are plosives (low frequency popping) on the vocals you can make a copy of the offending word and add some region specific EQ if your DAW allows, and then crossfade the new EQ'd bit back in, or use an automated EQ to duck out the low frequencies at the right moment.

It's also a good time to check phase relationships between any multiple mic sources, or layered kick drums or basses that have separate stems. Phase affects how the sounds react alongside each other and so you need to flip the phase of one sound while listening to both if you want to investigate the phase relationship. Sometimes there's an obvious cancellation of sound revealed, perhaps where some audio has slipped somewhere in the recording process, or because of a bussing or plugin latency issue. Don't assume that the files you have are correct! Sometimes regions can slip – mistakes can occur during the export / copy process. Sometimes it's a subtle thing, but make sure that you A/B with the rough mix to make sure it's sounding like it should. See the Using Reference Tracks box for tips.

Sometimes parts are just plain out of time. This is tricky. You don't want to be puritanical and lock everything to a grid (unless that's 'the thing'). You have to make a call on reining in the "humanity" in places if it's interrupting the flow of the music, or taking attention away from the top-line. It's OK to nudge a few bass notes here and there if they're not sitting with the drums. It's also possible to globally shift the timing of entire parts earlier or later by a few hundred samples to improve the groove.

You must try to be very careful here not to destabilise the feel. Sometimes a bunch of loose stuff sounds killer all together and rubbish if you "correct" it. So if you're not entirely sure then, for the sake of creativity, leave it well alone, and maybe re-visit for a little tweak when the mix is nearing completion.

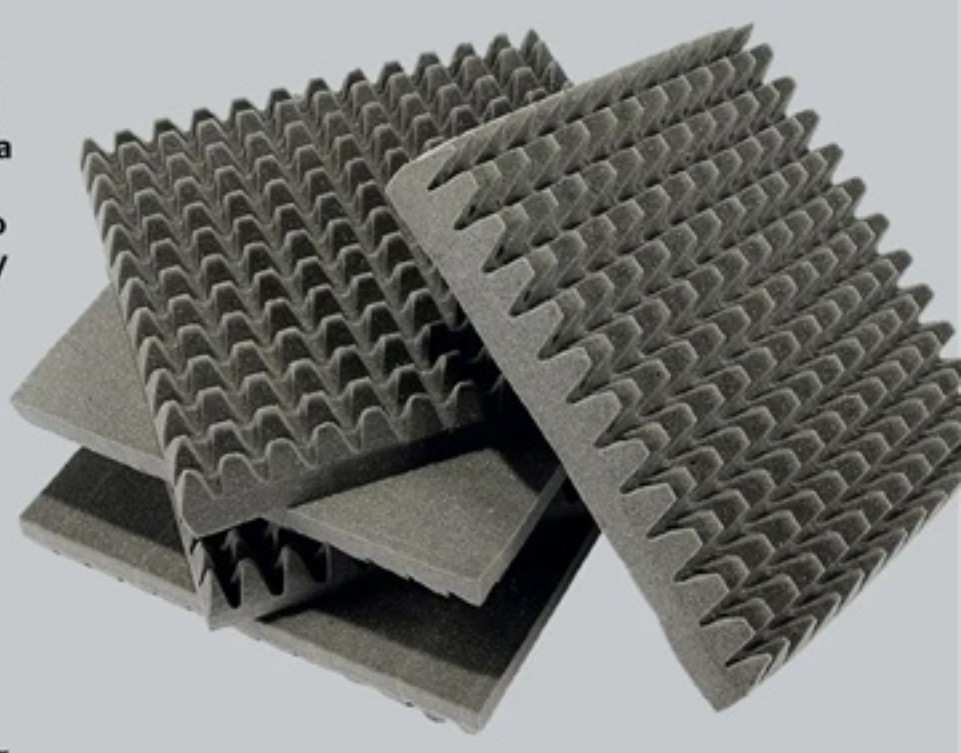
## Fix your mix room

Most people don't have good mixing rooms, let alone perfect ones. It's the combination of speakers and room that determine the sweetness of your sweet-spot. Most problems occur with low frequencies, but small spaces can also play havoc with your stereo imaging. Headphones, constant A/Bing and working at relatively low levels can help. But if your room is truly awful you should spend £500 on fixing it and £500 less on your monitors.

Ideally you want the sound waves generated by your speakers to travel uninterrupted to your ears, but all the reflected sound in smaller rooms interferes with, and adversely affects the direct sound. Your weapons here are absorption and diffusion, and you need to treat the surfaces nearest your speakers – including the ceiling.

Absorption prevents reflection. Effective bass (20Hz to 100Hz) absorbers are big and heavy with layers of moving membranes and varying high and low density material. By the time the low frequency waves have wobbled a membrane, sped up and slowed down a few times there's not much left to reflect back into the room, and what does has been sufficiently delayed in time to not interfere with the bass that's coming direct from the speakers. High frequencies can be absorbed using suitable foam (ideally Melatech) and heavy fabrics.

Diffusion is great for targeting high mids and highs (600Hz and higher), and redirects the reflections into a cloud that reduces their impact on the direct signal. There's no substitute for properly built wooden quadratic diffusers. Dennis Foley at Acoustic Fields offers much advice on the subject of small room acoustics, and you can also find BBC white papers and schematics for absorption boxes and bass traps online.



**No, gluing egg cartons to a wall isn't the same thing**

# The full mixing session

## Find a selection of accompanying audio files at filesilo.co.uk

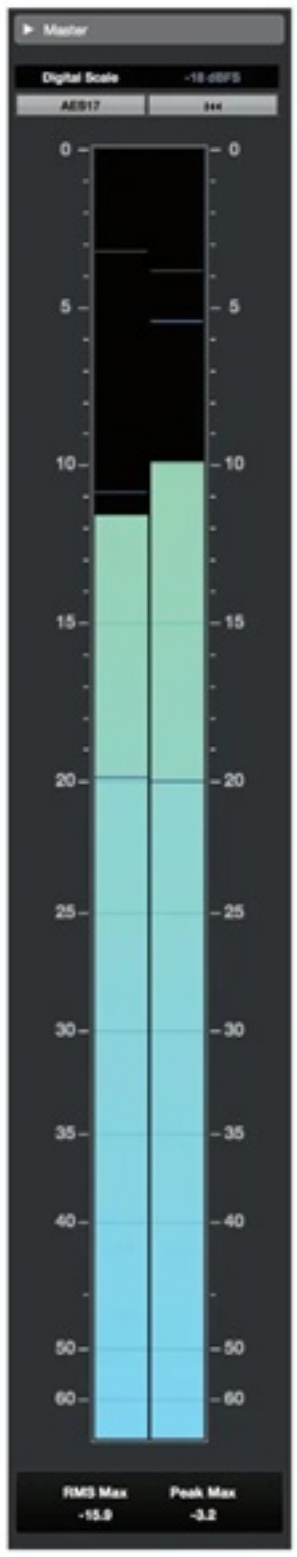
# Get a balance, set a level

Now is the time to begin the fun bit of the mix. We have to achieve a starting balance. This is where we set the faders at a position where the multitrack runs and everything is roughly in the right place. Don't expect this to sound in any way 'finished', it's simply a good point from which you can sculpt the final mix.

If you're mixing your own tune then you'll have got to a point where there's nothing major left to record and edit, and, as you have been tweaking away at the balance all along, it's probably not that far away from a good mix as it is. You don't have to pull all your faders down and start from scratch. You may need only a few adjustments here and there. On the other hand it may be that you've got into a mess and it's sounding cluttered and dull in which case starting again is probably a good idea. Make sure you have a bounce of your working mix to refer to, and keep your working session so you can go back and plunder settings from it if needed. Rough mixes have a 'vibe'. Don't lose it.

If you think you have a good balance, but your master fader meters are slamming into the red, then insert a basic level meter in the first slot of your master fader, solo your kick drum, select all your faders and pull them down until the kick is metering at about -12dB on the level meter. This will leave you headroom to add some healthy thump in the form of EQ and compression, and also allow plenty of room for your transients (momentary audio peaks), which is where the punchy, "hi-fi" goodness comes from.

If you're starting a balance from scratch then set your kick at -12dB and build your rough balance around this anchor point. This is the bit that needs practice and experience to achieve quickly. Just take your time and work at a medium volume. Try and find an average level for everything knowing that you will bring the mix to life with automation later on. We can't tell you how loud different elements should be, but if you feel you've lost the plot then check the "Starting Balance" box for some sanity saving tips and tricks to get you going.




## Creating headroom

Whether mixing a multitrack you've been sent or your own session, one thing is for sure: you need to check the headroom on your main output. Headroom is the amount of level available to you before your meters turn red. By default, our DAW output meters are set to 'digital peak' and will show a red 'clip' when you hit the loudest volume possible ie digital 0dB. Any signal louder than that will have its pointy waveform flattened or 'squared off', or 'clipped'. The audible result of clipped audio can be hard to hear when the clipping is minor, ie between 0 and 1.5 dB. But if you are attempting to squeeze too much level out of the master output then you will start to lose punch, high frequency transients or hits, and in extreme cases create actual growly, fizzy distortion.

Digital clipping is a type of distortion, and can be used to our advantage, but we do *not* want it happening on the master fader uninvited. Apart from it sounding bad it will cause you problems 'downstream' if you want to master, sell or broadcast your mix from a streaming service. This is a big subject worthy of its own article, but to cut a long story short: if your mix is too loud or clipped, a streaming service will turn it down and it may appear quieter than the music around it, which could well be extremely disappointing!

## Starting balance

**Achieving a basic starting balance for your mix can be a bit 'chicken and egg'. Mixing is basically about judging one sound in the presence of another sound. Go back to first principles; what you don't want to do is overload your outputs. Luckily, with most pop there's a kick drum rooting the sonic picture, and we can use this to help build safe headroom into the mix. When the kick's in place, setting the snare and bass becomes easier. Put a basic level meter over your master bus, solo the kick drum and set the kick drum fader so the signal at the master outputs is peaking at -12dB.**

**When really struggling you can try the pink noise trick to help get the different elements of the mix to some kind of tonal balance. After your kick drum is set, create a new channel and insert a pink noise generator, solo that too, and set the level of the pink noise so you can only just make out your kick drum. One by one, introduce the different tracks and turn them up until you can only just make them out among the pink noise. As the pink noise is generating even, broadband noise, you end up with a listenable balance to get you going.**

d16 Group Frontier limiter can be found in the CM plugin suite

# Compression - the glue

**From the cm collection, Ignite VST Vice One and eaReckon CM Comp 87 are just two of many compression choices**

Hopefully you have a good starting balance that is far from clipping the master outputs. Chances are that it will still be sounding a little bit flat, and is in need of some "va-va- voom" and "je ne sais quoi".

Successful mixes have a quality where the different elements affect each other in a complementary and natural sounding way, creating what's often called a 'glued together' mix. This is achieved using subtle and carefully applied compression.

The tools we have for constructing a mix fall into The Big Five categories: compression, EQ, delay, reverb, and distortion. Each of these categories have hidden depths and, judging by the amount of questions asked, the least well understood of these is compression.

Compression, or dynamics control, comes in many and various forms but always does the same job. It automatically adjusts the volume of the sound being compressed. Often the desire is to create the illusion of a loud sound at a low volume. A recording of some drums coming out of a Bluetooth speaker will never be quite as loud as someone playing a kit right next to you would be, but we use compression to mimic how the ears and brain deal with loud sounds. Your brain adjusts to incoming levels in real time to let you hear the info most effectively. So does a compressor.

You set a compressor to turn the volume of the signal down when it reaches a defined volume measurement (threshold), by a certain amount (ratio), happening at a certain speed on detection (attack), and letting it get back to normal over a certain time (release).

> "The tools we have for constructing a mix fall into The Big Five categories"

A 2:1 ratio means a signal that's 2dB over the threshold will be turned down so only a 1dB increase happens. So a big ratio like 10:1 means a sound that's 10dB over the threshold is turned down by 9dB. The volume level is squashed. But 'squasher' didn't stick, so 'compression' it is. Analogue compressors are not perfect, like digital compressors, and the various ways in which they fail to turn things down fast enough gives them identifiable, 'musical' characteristics.

Knowing what type of compression to use when takes experience and much critical listening. Many golden-eared pros can tell you what compression they are hearing, particularly with analogue compression during recording, even down to make, model and settings! You need to grasp the basics of this too. The best way to do this is to watch a gazillion examples on Youtube and mess about in your own sessions. We'll use different types of compression to make the quiet parts of a vocal louder, or make the drums sound punchy, or to 'hold' the mix in place.

## Submixes and masters

Before you start treating the individual tracks, get an idea of how the multitrack hangs together as your starting balance, but bussed out through some nice sub-mix and mix bus compression. We'll use a simplified version of a popular bussing setup perfectly suited to fully 'in the box' mixing. We'll create five submix busses: A) drums and percussion, B) basses, C) other instruments, D) vocal and E) Clean. All our tracks will be routed to their respective busses and these four busses will then be routed to our main master bus. Each of these busses will have their own compression and EQ.

Bussing the multitrack this way means you can still make big alterations in specific areas right up to the last minute without upsetting careful balancing elsewhere in the mix. For example, hours spent crafting the balance of the mix could be undone by cranking the kick drum up a few dB at the last minute; the kick will hit the compressor harder, and the gain reduction will increase, and the lead vocal could be pushed back into the mix, and other elements could end up sounding less present. You'd then have to rebalance everything else to accommodate this hike in the level of the kick.

Having separate sub-mix busses makes it easier to adapt to changes and control the level hitting the master bus so the mix can breathe. A mix has the impression of being 'glued' together because of the compressors inserted on the sub-mix and master busses. Often these compressors will only be tickling the levels out of their outputs, perhaps with no more 2 to 3 dB of gain reduction each, but the end result sounds like a party in a big mixing console, and not a bunch of disconnected sounds sat alone in the digital equivalent of a waiting room.

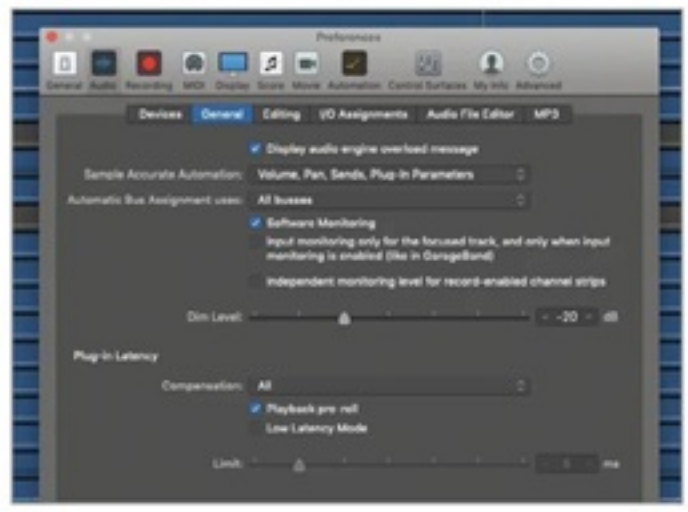
The free and rather good **cm** Plugins available to you include a bunch of different flavoured compressors suited to different jobs. One of our faves is the Frontier from d16 Group. This is what we've put on our master mix bus to help with the glueing process. Frontier has super simple controls and automatic gain make-up which may not be to everyone's taste, but we like it as it sounds 'musical' and with just the release time and threshold to play with you can get where you need to quickly. On the mix bus

**SK Note Snap for transient shaping**

## > Step by step

## 1. Bussing - subs and masters

1 > The I/O buffer size governs how your CPU deals with a pile of audio. In terms of a theoretical bucket chain, a big buffer is big buckets occasionally, and a small buffer is small buckets constantly. Low buffers offer low latency – good when recording, high buffers reduce CPU strain – good for mixing. Look in your DAW's audio preferences.



2 > Each plugin or aux bus you use has an inherent latency. Your DAW's latency compensation looks for the longest processing delay, and adjusts the playback timing of everything else to ensure you hear the right sounds at the right time! These calculations change as you use plugins, which is why you need automatic delay compensation ON.

3 > We need five submix busses. In Logic these are created in the mixer window using the 'Create New Auxilliary Channel Strip' from the Option drop down menu or the 'alt cmd N' key command. Our five new submix busses are set to inputs 26 to 30 respectively and named 'A Drums', 'B Bass', 'C Band', 'D Vox' and 'E Clean'

4 > To prevent our multitrack channels going straight to the main mix outputs we select multitrack and affect return channels (not the five submix busses) and change their output to Bus 31. This creates a Bus 32 mixer channel which is labelled 'Zero" and turned down completely. Now when you hit play you wont hear anything, and our multitrack is ready for bussing.

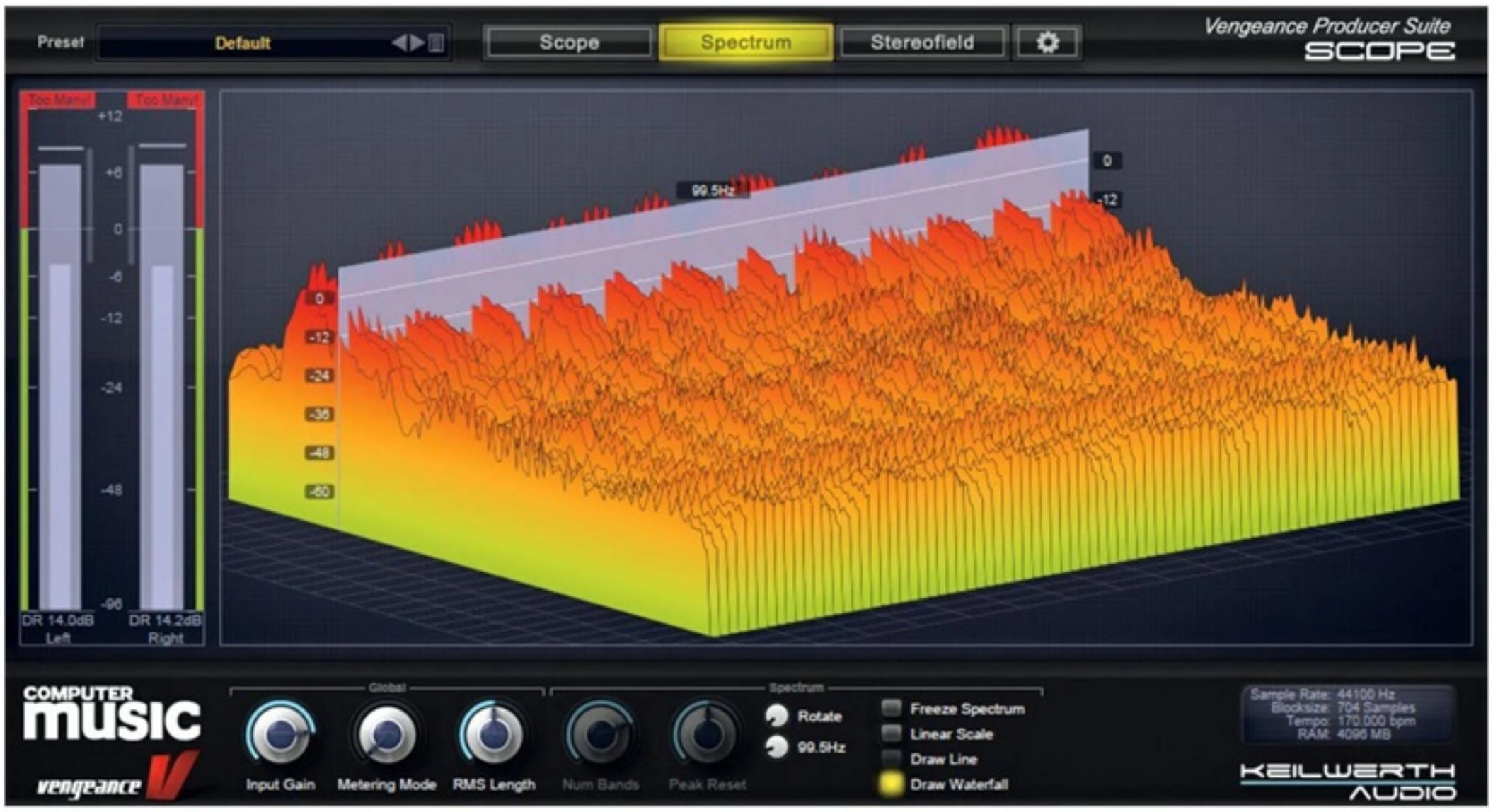
5 > To stop getting confused at this point it's worth taking a minute to get organised. Group your multitrack channels into categories and colour code the regions, tracks and mixer channels accordingly. Here we've broken down to drums, bass, band and vocals. You could try drums, sustained sounds, percussion, vocals, guitars, keys... whatever works for the mix.

6 > Now select all your drum channels on the mixer page and add post-fader Aux sends to Bus 26 or 'A Drums', and set the send level to OdB. Now select the bass and send to Bus 27 "B Bass" etc. These groups of sounds will now be audible through the master mix bus via our new submix busses.

7 > Each submix return can have different treatment. Having the fifth clean bus offers an additional parallel path to help something cut through any submix compression if needed. We've inserted the Satson CM channel emulator over all five busses including the clean one to add another layer of subtle harmonic distortion in our quest for 'warmth'.

8 > On the 'C Band' and 'D Vox' busses we're using the CM-COMP87 and PTH2A-CM EQ. Here's a new CM plugin for the 'A Drums' bus. SNAP CM Edition from SK Note is a deceptively simple but unique dynamic EQ which changes the tonal characteristics of the 'HIT' and 'BODY' parts of the sound. Set like shown it indeed makes the drums 'snappier'!

9 > Another free **cm** plugin we've not seen yet is the Ignite VST Vice One compressor. Here we have the option of a more coloured 'Analog' sound and a 'soft-knee' for a gentler onset of compression. A final 'Saturation' control allows further harmonic distortion, always a winner with basses. The compression is gentle here. The tone shaping is more important.




**Keep an eye on output with the CM Vengeance Sound VPS Scope**

we're setting the threshold high so the compressor doesn't kick in till things get properly loud, and consequently seeing just a tiny bit of gain reduction at the loudest points. But that's enough to help gel the overall sound. We're also taking the output level down a bit so we don't clip the input of the next plugin in the chain, which happens to be an EQ, and leaves us room to boost any frequencies we like without clipping the master outputs. Always keep headroom in mind.

Eventually everything goes through the master compressor, but on the way there our different sounds are treated by different compressors on each submix bus. On the drum bus is Snap, more of a transient designer than a tradition analogue compression emulator. On the bass bus: Vice One, set to analogue on a low ratio, only really kicking in when the two smaller staccato basses happen alongside the drone bass. The 'Band' and 'Vox' busses both have our Comp 87 set on a low ratio with fast attack and release times, a 60/40 dry/wet mix and never more than 5dB of gain reduction.

The **cm** Vengeance Scope plugs in the last slot of the mix bus helps us keep a visual check on output level. Setting up sub-mix busses can be daunting, but have a look at the Bussing – Subs And Masters box for help with how to achieve this essential step into the world of grown-up mix engineering.

Elsewhere in the mix Outlaw CM is keeping the picked Nylon Guitar in focus by bringing up the quieter bits. HoRNet FatFet is emulating a classic Urie 1176 compressor and flattening the attacks of the Wurli electric piano part to keep it smooooth and silky.

## Make it worse to make it better – distortion

Beautiful distortion, another of The Big Five mix tools, helps bring a mix to life. Imagine Daft Punk or Justice without distortion, or Jimi Henrix playing a clean DI'd Strat? Actually he'd have made it work to be fair. In this mix we have'nt used much in the way of obvious distortion. The vocal stem has a little baked in crunch of its own, otherwise we certainly would have been blending in a little saturation to help the lead vocal take its rightful place front and centre of the mix.

Let's take a closer look at our submix busses though. As well as their compressors, they each have a Satson CM saturation plugin, emulating gentle 'driven mixer channel'-style harmonic distortion, which subtly seems to add a bit of 'life' to the sound. The quality added by this type of harmonic distortion is hard to define; the sound seems a little brighter in the mids and a little more forward in the speakers.

As with compression, the trick is to use a little in a lot of places as opposed to one big dollop on the mix bus. Though we do have one on the output bus too. A sprinkle not a dollop.

**Outlaw CM compressor and Satson CM channel strip emulation for saturation**

# Working the lead vocal

We're about halfway through the process. We have a basic balance that's sounding glued together in a good gain structure. The next priority is the Lead Vocal. Every mixer will have a different approach to this vital element, and the 'vocal chain' used by revered mixers is the stuff of industrial espionage. But here's the thing: you have to respond to the source material. Every sound recording is unique, and an EQ preset called 'Ace Lead Vocal' might sound ace, but these presets, and other people's techniques should be looked at as a possible starting point for your own judgement. How you think a lead vocal should sound is what's going to make your mix your mix.

The rules are: the lead vocal sits clearly in the mix and you can hear the words, or made up words, or sounds, and at the same time it's part of the music, not stuck on top like a voice-over. In addition, you have to keep the vocal interesting sonically throughout the track.

**The Microfuse reverb can help you with 'blip' management**

Subtle variations in treatment and careful use of backing vocals will help you achieve this. The lead vocal for verse one of *Morning Light* is a single vocal. Ours happens to have two compressors, two de-essers and an EQ. See the Lead Vocal Chain box for the details. We're also sending on an aux send to our Reverberate CM reverb plugin for our basic sound. Lead vocals are rarely bone-dry (except for a special effect), and a well balanced reverb helps the vocal 'sit' among the other sounds in the mix. You may find you need to automate the send level to your main reverb as the instrumental backing becomes more or less loud and dense.

A double-track is where the singer duplicates his performance, not by copying and pasting, by singing it twice. The variations between the performance create a natural chorus-like effect.

## > Step by step 2. The lead vocal chain

1 > The CM-EQUA 87 from Eareckon is a good choice from **cm**'s free plugin collection for general EQ duty. Its slopes are fairly gentle and you get one Cut band and three parametrics. In this case we're subtractively EQing some bottom end from the lead vocal from about 220Hz to effectively brighten it a little without adding top.

2 > Logic's own compressor is inserted as phase one of a double compression approach. This stage uses fast attack and release times to heavily compress the vocal, but the Mix control is set to around 50%. This keeps the sound more natural, but has the effect of making the quiet bits louder, effectively reducing the dynamic range of the signal.

3 > Gain forming part one of a two part process the TB_Sibalance_CM from Toneboosters provides the first stage of de-essing. This brings down the sibilant or 'essy' sounds which can poke out, especially if more top end is added later. Each de-esser will have its own characteristics and be better at targeting certain types of harsh sounds.

4 > Now the level and sibilance have been partially addressed we can use OverTone DSP's EQ CM to properly attack the tone of the lead vocal. This Pultec style emulator can pile on the top end in a way that is instantly recognisable. In this case around 10kHz. We also add a little bottom end back in around 100Hz for warmth.

5 > The CM-COMP87 offers gentler control, now that the dynamic range has been reduced already. The attack and release times are still fast, the ratio's about 6:1 and the mix control still allows some of the uncompressed signal through. This vocal will have been compressed six times by the time it gets to the final mix.

6 > Last in the chain is our second de-esser, this time Logic's own DeEsser 2. All the compression and EQ is done now so this de-esser has the last word in taming sibilance, and works with the earlier one to ensure all bases are covered. We found we were tweaking both of these de-essers right up till mix time.

"These are the kind of tricks you need up your sleeve to keep the LV interesting throughout the track"

In this mix the double track comes in for each chorus and helps the LV hold its ground against the backing track. We've compressed the double-track quite heavily with Comp 87 and used the excellent PSP cmDelay to create a short delay on either side, of different value. This makes the double-track appear to sit either side of the LV rather than bang in the centre. We also have a double-tracked vocal harmony which comes in at various points. These have been compressed, panned at about 50% width and sent to the same reverb as the LV.

Where things get softer in verse 2, which feels more like a break, we have automated the ratio of the Comp 87 compressor so the LV sounds more natural. It doesn't need the heavy compression to poke out as there's not much around it. For the ad-lib 'oohs' after each chorus we've duplicated the LV track and copied these bits to their own track and used EQUA87 to filter the bottom end out and used a much longer 350ms delay. These variations and automations can be big, but not noticeable, or designed to make a sonic impact, like the long delay. These are the kind of tricks you need up your sleeve to keep the LV interesting throughout the track.

Using the vocal reverbs and panning the BVs helps begin the process of creating the imaginary depth to your mix. Some on the early synaesthesia spectrum will also talk about 'height' too! Pick your moment for big reverbs. In this mix we've made the occasional blip sound very 'verby with the Microfuse reverberator. In fact we've even edited a few of the blips out to create more space.

Sometimes a short 16th or 8th beat delay can really help a sound appear to sit in a space without the constant tail of a full-on reverb. **cm** plugs like Sidewidener and Panagement help 'place' the nylon guitar and one of the Wurlitzer electric piano tracks into some interesting stereo spaces.

**Place instruments in the stereo space with Panagement CM**

## > Step by step 3. Going deep on the vocal

1 > To add a general reverb to the main vocal a post-fader send on Bus 1 is created in slot 1 of the LV channel. This in turn creates a return channel for Bus 1 in the mixer and Reverberate CM plugin is inserted in the first insert slot and set to 100% wet. Adjust Bus 1 send level to taste!

2 > The "oo-oo" sections of the LV need a different treatment. This really helps keep the LV sonically interesting through the mix. The LV track is duplicated to preserve the plugin chain, and the "oo-oo" sections are cut and moved onto this new track. Now we can give these parts their own sound and level.

3 > The de-esser plugin on for this "oo-oo" channel is bypassed as there's no sibilance, and we want all the high-frequency content intact to help the oohs sound breathy and ethereal. The PSP CM Delay is inserted at the end of the chain and set with a 350ms ping-pong delay with moderate feedback to create an obvious and spacious effect.

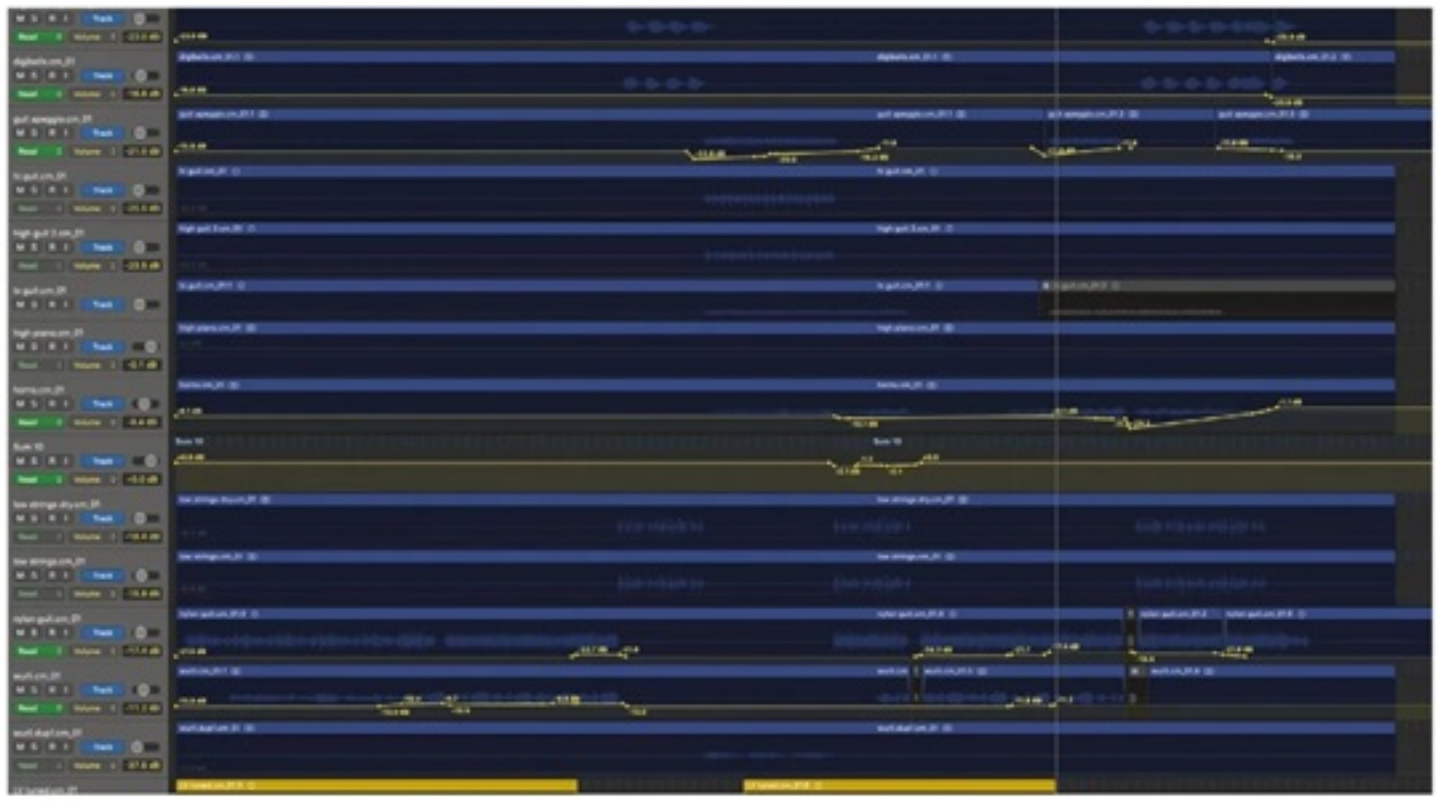
4 > During the main body of the track our heavy compression on the lead vocal is great, but during the break it's a bit heavy handed. To make the vocal sound more natural and relaxed at this point we can automate the compressor ratio. The 'CM-COMP 87 Ratio' is selected from the LV channel's automation drop down, and adjusted to 4:1.

5 > Now the level of the main vocal is automated throughout the song. 'Volume' is selected from the automation drop-down in the LV tracks and small adjustments are made to sit the LV exactly right against the track, and some bigger rides are made to calm down any syllables that pop out or disappear a bit.

6 > To help the LV achieve the elusive effect of 'floating' in the middle of the overall soundstage we decided to set up an aux send from the LV to our 'Clean' submix bus which is on Bus 30. This creates a parallel compression setup, and only a small amount of LV is needed to bring it 'forward' in the mix.

# Top ‘n’ tail – especially top

Don’t be afraid to make any part of the song better with some judicious cut ‘n’ paste if it seems easy and obvious to you, especially the intro. Clean it up and pay very careful attention to the energy as the song opens and different sounds arrive. There’s a sweet spot for how loud things should be before the main body of the song kicks in. Too loud at the intro and the main groove feels like a let-down. Also pay attention to the end. In our case we copied the sparkly guitar arpeggio and nylon guitar part to make a better ending than just leaving the digital bell part hanging as it was. Always save an edited version though – in case your talent is ever unappreciated!



## > Step by step 4. The mix bus chain

1 > We set up a basic level meter in the first plugin slot on our main output fader or ‘mix bus’ before anything else. This will help keep a watch on levels into the mix bus and make sure that no unwanted clipping is happening into the next plugin. If you’re in the red then turn down your mixer or submix channels.

2 > Next up is the SatsonCM plugin from Sonimus. You can see from the input level that no additional gain is need here. This plugin emulates classic mixer style harmonic distortion. This generally means a subtle boost of second order harmonics and adds a little ‘life’, and begins the process of gently clipping of the very loudest transients.

3 > SunRuysCM is a characterful compressor with basic control over a signal’s dynamics. By bringing the Threshold down and the Ratio up, we start to get a ‘pumping’ volume reduction effect on the audio. Bringing the Makeup parameter up a bit to compensate for the gain reduction, we can tell we’ve ‘brought out’ the sound, making it more prominent.

4 > Next is the OverTone DSP’s EQ CM. This emulates the classic Pultec EQP1, famous for being able to successfully add hefty amounts of bass and treble to the whole mix if needed. Here it’s subtly, but effectively, adding a little 30Hz right down the bottom and some 10kHz at the top, as well as its own gentle harmonic distortion.

5 > TB_Barricade_CM from Tone Boosters is the last in our chain and brings a mastering grade finishing touch by reducing the peakiest transients by using superfast attack times. Used gently this should not be noticeable and help you hit a good perceived loudness, but also keep a sensible dynamic range of around 10dB in your final mix.

6 > The VPS Scope (CM) from Vengeance is there to analyse. Set to ‘Stereofield’ we see the Goniometer showing us phase information. That green bar must stay as much as possible on the right hand side. Too much on the left and you’ll have problems cutting vinyl! The loudness meters on the left side are your final loudness measurement.

# Using reference tracks

If there's one tactic guaranteed to fast-track your mixing and mastering skills, it's comparing your work-in-progress to great-sounding reference tracks. When done correctly, you'll be able to identify deficiencies in your mix compared to those pro songs; evaluate areas of improvement in terms of frequency, dynamics and stereo balance; and formulate processing strategies that will not only improve your current song in line with world-class records, but also elevate your long-term mixing abilities. Referencing (or 'A/Bing') can be done directly in your DAW. However, this process can become frustrating as you battle with multiple song setups, level inconsistencies, file formats, playback positions, fiddly routings and so on. Thankfully, a handful of dedicated referencing plugins have cropped up in recent years, with features ranging from simple back-and-forth switching to more extensive analysis, complex signal tapping, and even blind testing.

> Step by step

## 5. Controllable sidechain duck on bass drone

1 > There is a soft and a hard kick drum in this session each on their own track. To create the sidechain we need to open a pre-fade aux send on each channel set at 0dB. Its own bus return channel in Logic's mixer is created and named sc1, which is turned all the way down and solo isolated by ctrl clicking the solo button.

2 > We insert Logic's own compressor on the 'low drone' mixer channel, and select the send source used on both the kick drums, in this case Aux 32. We now select the Side Chain option in the compressor itself just below. The bass drone will now duck with each kick drum when the threshold is lowered. Adjust ratio and release to taste!

3 > A CM-EQUA 87 is inserted. The cut band is set to LO CUT at 52Hz. The slope of this filter is quite gentle and takes the lowest subs away. The MF and HF bands are used to boost the upper mids and tops to drastically brighten up the sections of bass where the filter has been opened during the recording.

4 > Where the bass drone is present with no kick drum, we still want it to pump. We're not using a fancy plugin for side-chain effects so we create a side-chain source for these moments. A new track is created called sc trig extra with the same Bus 32 pre-fade send as the kicks, and the volume set to off.

5 > A four-bar section of the small kick region is copied and pasted onto the new sc trig extra track at the points where the drone bass is playing without either of our two kick drums. Once at the beginning of the song and again a bit later during the break section.

6 > Controlling the amount of side-chain ducking on the drone bass happens by automating the amount of kick drum sent to the compressor via Bus 32. From the Automation drop down menu on the sc trig extra track select "send 1 sc 1". Now the level of the kick being sent is automated to make the ducking more subtle.

# EQ – what and when?

What is 'flat' sound? Who's to say we all hear the same or see colours differently? We will never know. When we record we select microphones to suit the source. Often that means we know certain mics have a sensitivity or tendency to reject certain frequencies. That's how we avoid tinny cymbals, scratchy acoustic guitar, sibilant vocals or kick drums with no kick. By selecting different mics to record, or setting the cut-off frequency on our synth we are making EQ (equalisation) decisions.

These sounds speak differently in a mix. As a mix engineer you have to decide what element of the sound is saying the most, and what parts of the sound could be dispensed with. Do you need the very low end or the very top? All that energy mounts up with every different track and can create the dreaded 'mud'. Which EQ for what purpose though?

For surgical corrections like an exaggerated ring in a drum, or a boomy honk of a room, you need a tight Q (the Q is the ratio of the centre frequency to the bandwidth – or – the pointiness of your peak or trough). Use the Overtone DSP AF2-10 CM or DDMF CM EQ Pack or your own DAW's standard EQ for this surgical stuff.

For broader tonal shaping you need a wide Q or a Shelving EQ, which cuts or boosts everything above or below a chosen frequency. For your master bus you should try a 'program equaliser' like our PTH2A-CM Pultec EQP1 emulation. This acts similarly to the 'tone' controls on a stereo, and allows broad bass (LF) and treble (HF) boosts that sound 'musical', not targeting narrow areas of the mix.

As a general rule you won't need much action below 30Hz on any individual track. Don't be afraid to cut and boost super lows and highs, but use high-pass and low-pass filters (extreme shelving EQs with steep slopes) to lose unnecessary high and low clutter.

Sometimes boosting the most active part of a sound is an alternative to simply turning the channel up. Always try and EQ a sound in the context of the whole mix if you can. No-one's going to listen to it in solo!

**Overtone DSP AF2-10 analogue filter modelled EQ**

**The DDMF CM EQ Pack – two superb equalisers comprising IIEQ Pro CM including 19 filter types and the LP10 CM, a linear phase mastering EQ. Below – Overtone DSP's Pultec emulation**

**Below - Overtone DSP's Pultec emulation**

# Monitor and room correction systems

The impact of 'coloured' speaker or room on the mix is an old problem. 'Neutral' sound has always been the holy grail of acoustic designers. If your speakers or room were too bassy, your mixes came out not bassy enough, because they sounded correct to you when you were mixing them. You can put up a mic in the mix position, and watch the level meter of your DAW move as you change the frequency of a test tone played through the monitors. Ideally it should stay flat, but usually doesn't (see *Your Mix Room*).

We now have affordable diagnostic tools to analyse our frequency response, phase, distance and even height! Ranging between £150 and £10,000 plus, there will be a solution that fits your set-up, and the entry level systems can work very well.

IK Multimedia ARC System 3 offers a software and calibration microphone package enabling you to insert a correctional plugin within your DAW to flatten your curve, and even simulate other listening environments.

Sonarworks Reference 4 Studio goes a step further and allows you to insert their correctional voodoo 'systemwide' to allow you greater flexibility between devices and software. Slate are here with their VSX-calibrated headphones, and more are joining them.

The Trinnov ST2, which comes with a multi-capsule '3D' measurement mic with four outputs, is standalone hardware with comprehensive connectivity, high end converters and basic monitor control. At the very top of the tree are Trinnov's D-Mon12 and Avid's modular MTRX controller, which not only do all the analysis and correction for full-blown multi-channel immersive audio, but also offer full studio monitor control and routing facilities.

**Sonarworks Reference 4 Studio (above) and the IK Multimedia ARC System 3 (below)**

# 10 quick mixing tips for the road

## STAY IN CONTEXT

**Listen to other material while mixing. Listening to a subby bass in solo makes your mix sound 'toppy' for a while when you jump out of solo. Try to EQ your sounds in the context of the full mix, and keep bouncing around between other good mixes to keep your sonic perspective.**

## WATCH YOUR STEREO

**Beware super-wide sounds. If the mix works in mono you're winning. Hard 100% panned left and right guitars sound great on speakers but can be distracting on headphones – try 80%. If a phase correlation meter on the mix bus is moving left of centre you've gone too far. Extreme stereo works, but choose your moment carefully.**

## MIX INTO YOUR MIXBUS COMPRESSION

**The plugins across your mix submix busses and your main output mixbus should be applied early on in the process, pretty much as soon as you have a good starting balance. A good glue from submix and mixbus compression goes a long way towards a final result, don't try and achieve the final mix with only each channels individual treatments.**

## TURN IT DOWN

**Blasting a mix in the early stages is fun, but that's because your ears are adding some great human compression to the mix and those pesky Fletcher-Munson curves are skewing your perception of the mid-range. Going from loud to quiet is also problematic. If the mix sounds exciting at medium monitoring levels then it is.**

## DO YOU NEED IT? DE-CLUTTER...

**Sometimes there's too much stuff. Maybe you don't need it all on all the time. If low level stuff is hidden, try turning it off if it's not adding anything useful. Explore quiet parts – there could be a genius hook buried there. Every part should be scrutinised and has to earn the right to be in the mix by doing a necessary job.**

Boz Digital Labs Sidewidener is great – with caution

## TIDY UP

**When you have a stack of vocals in your mix, or lots of mic recordings there will be areas of quiet. In those quiet bits can be breaths, string noise, hum etc. Add all this together and it can get surprisingly chaotic. Cleaning the tops and tails of vocal parts and guitar entrances and exits can bring a perceptible clarity to the overall mix.**

## ALIGN YOUR SIBILANCE

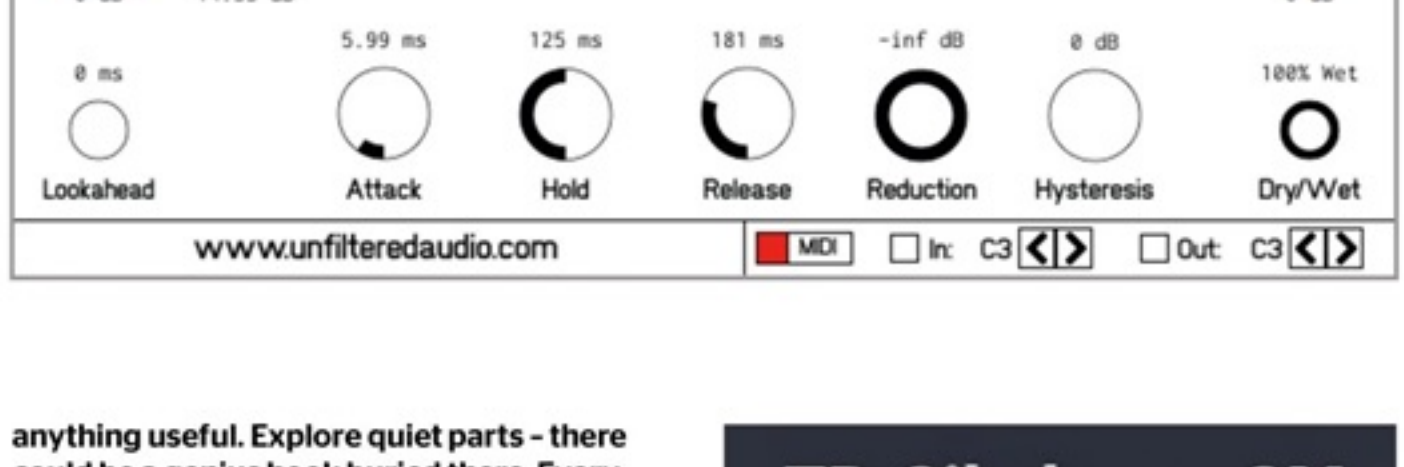
**Small differences in phrasing from different singers, particularly with stacked harmonies and double tracked vocals, can turn S sounds into a real mess. You can manually edit the S sounds into line with the lead, turn the Ss down, or even mute them out completely. This can make your backing vocals sound super locked – great for pop.**

## DON'T COMPETE WITH A LOUD ROUGH MIX

**Rough mixes are often printed with a loud limiter 'for vibe'. If you're referring to the demo, which is often a great idea to help preserve the intent of the track, turn a limited rough mix down by 6db or so. Loud always sounds better, so level your playing field and don't worry about final, mastered level.**

## SNARE AND VOCAL LEVEL

**A golden rule, agreed on by mixers of quality pop, is that when you turn the volume of a good mix gradually down to nothing, the last things you should be able to make out are the lead vocal and the snare drum. The lead vocal can be surprisingly 'back' in the mix up loud, and still be right.**



## GET ORGANISED

**If you're working for someone else, keep a sheet for each title that shows you're happy with the multitrack you've been given, when it's due, what is due (versions, stems, filetypes etc). Note feedback points and keep track of when they have been addressed. Have a good system for version tracking so you can easily refer to previous mixes. cm**

# 10 quick tips: width + depth

Getting extra dimensions in your mixes can really give them a pro feel. Here's how...

01 Pitch and formant shifting is handy for quickly widening mono synth, vocal and guitar tracks. Create two copies of a track, pan them hard left and right, then apply slightly different pitch shift or formant shift values to each copy. Duplicate additional tracks with alternative pitch and formant settings for an even wider effect.

02 To widen a stereo track, flip the reverb or delay so that the effect from the right channel is played in the left and vice versa. Set up a 100% wet reverb or delay effect on a buss and then flip the L/R channels using a utility plugin.

03 Try rotary plugins and autopanning effects on mid-/high-frequency percussion-like hi-hats, shakers and bongos, giving the tops width while the kick and snare stay dead centre.

**Lower the highs and boost 200Hz to add distance**

**Autopanning plugins work particularly well on hi-hats, creating movement in the high frequencies – use subtly**

04 The human ear can make out only a few different panning positions at the same time. Use this knowledge to keep the panning of individual tracks nice and simple. Using just five panning positions will keep your mix clean and increase separation between instruments. The positions hard left, mid left, centre, mid right and hard right can be used exclusively, so don't be too scared of doing hard panning.

05 Collapse the majority of a track's elements to mono to enable the stereo and panned instruments to really stand out. Alternatively, use a multiband imaging plugin to switch all the frequencies below a certain range into mono. This will allow the midrange and high frequency instruments to become more prominent.

06 To create a wider-sounding guitar riff, take a mono guitar sample and duplicate it onto three separate tracks. Pan one left, one right and one centre, then apply guitar cabinet plugins to the three tracks using different settings on each. Keeping a 'clean'-sounding centre channel with two different distorted cab emulations panned left and right sounds especially good.

07 To make the chorus parts of a song really stand out, widen the stereo field during that part and narrow it again for the verses. Use a mid/side plugin with an adjustable width setting on wide-sounding instruments. Then automate this width setting to reign in the stereo effect during the verses and widen during chorus sections.

08 Careful EQing is one of the simplest ways to create an impression of depth in a mix. For sounds you want to place 'close' to the listener, set the EQ curve to brighten them, exaggerating their dominant frequencies. For 'farther away' sounds, roll off the top-end around 2kHz and slightly boost the low frequencies below 200Hz.

09 It's easy to create a basic perception of depth with just one reverb instance to save on CPU and keep your mixer routing simple. Firstly, set up a buss with reverb set to 100% wet. You can then vary the send's level depending on how far away you want each track to sound.

**Automate mid/side levels with a plugin like Voxengo's MSED to make your chorus stand out against the verse**

## 10 The human brain is rarely able to pinpoint more than two or three positions of distance simultaneously. We can use this knowledge to simplify the depth-staging of our mixes

**1 >** It's really difficult for the human ear to make out more than three levels of depth. Here we'll use reverb plugins to mimic three depths of a virtual soundstage and then buss tracks to them. We're starting with a basic tune consisting of just seven tracks.

**2 >** We then set up three buss channels to represent different distances: near, mid and far. One for the beats and bass (close), a second for guitar and sax (mid) and a third for the piano and organ (far). We then add the same Live Club preset reverb to each buss.

**3 >** Now on each of the three buss channels, we change the reverb's Wet level, Diffusion and reverb Time. We dial them back for a closer sound for the beats and bass buss and increase them for the piano and organ buss to push them to the back of the mix.

Download the accompanying video for this feature from here:
**filesilo.co.uk/bookazines**

# Adding width to a mix

Are you battling a mix that includes too many mono sounds? Maybe they just need a bit of added width. Welcome to our first Mix Masterclass...


## Jon Musgrave

**Jon is a London-based producer, mixer, composer and club remixer with a diverse CV that spans dance, pop, rock and music for media. He's also a long term contributor to *Computer Music*. Whether working alone or collaborating he usually handles final mixdowns, which is why we've got him to share some pearls of mixing wisdom.**

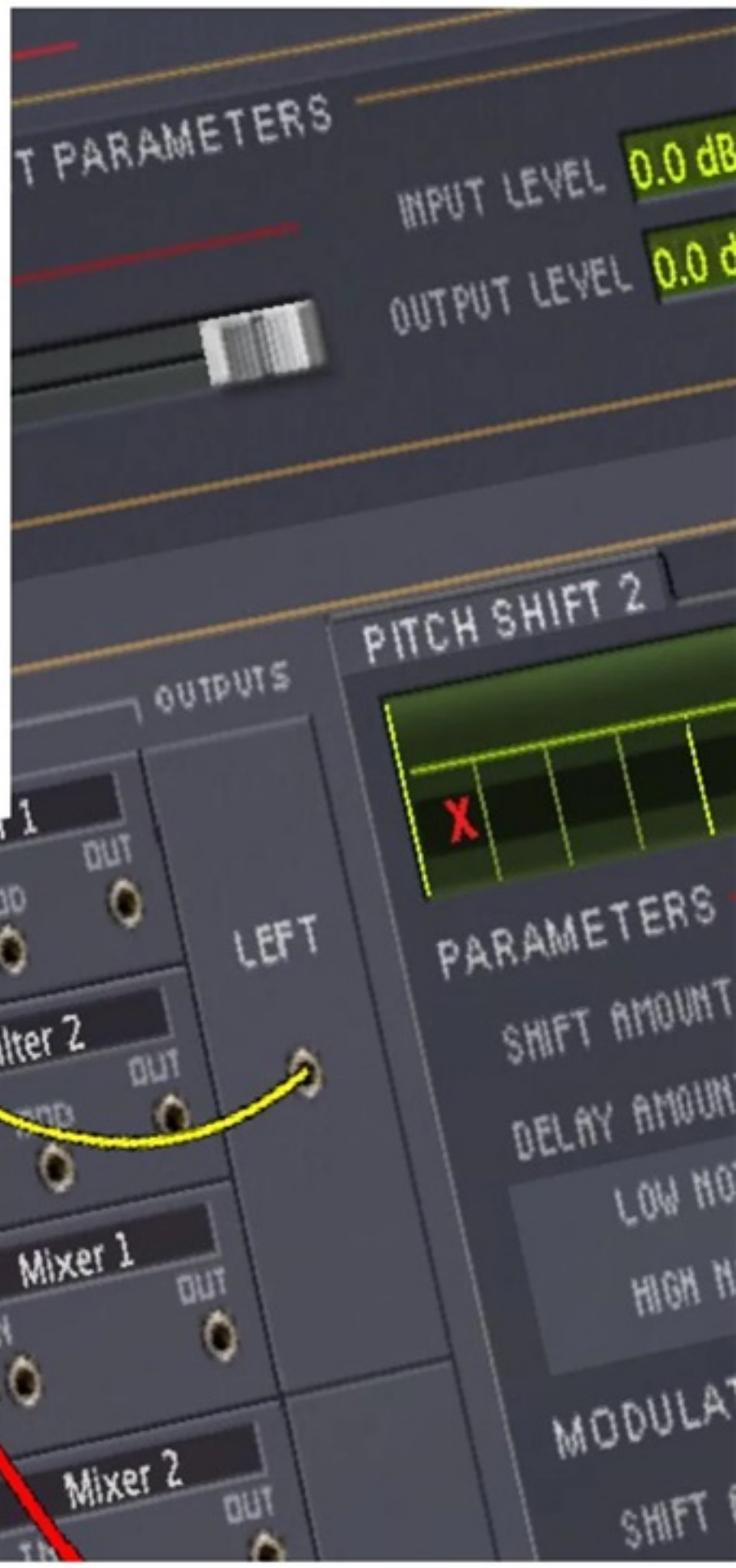

**> Making full use of the stereo image can be an important aspect of a successful mix, and often we need to find a way to add width to mono sounds.**

The groundwork for a wide, expansive mix is often laid at the production or recording stage. You might have double tracked elements to create really tight doubles that you can pan to opposite sides, or maybe you've recorded instruments in stereo complete with a great room ambience. Either way, these components mean you're already well on the way to stereo nirvana and can supplement these aspects with further reverbs and panning to create stereo width as required. But what if you don't have these elements in the first place and are presented with a bunch of mono sounds?

An obvious solution is artificial reverb, but this adds a lot of colour. Here I'm going to look at a number of ways we can add width to mono sounds without that reverb colour. These techniques are pretty straightforward and use some form of delay, pitch shifting or pitch modulation. That said, they can also have downsides, particularly when you mono them, which we'll also discuss.

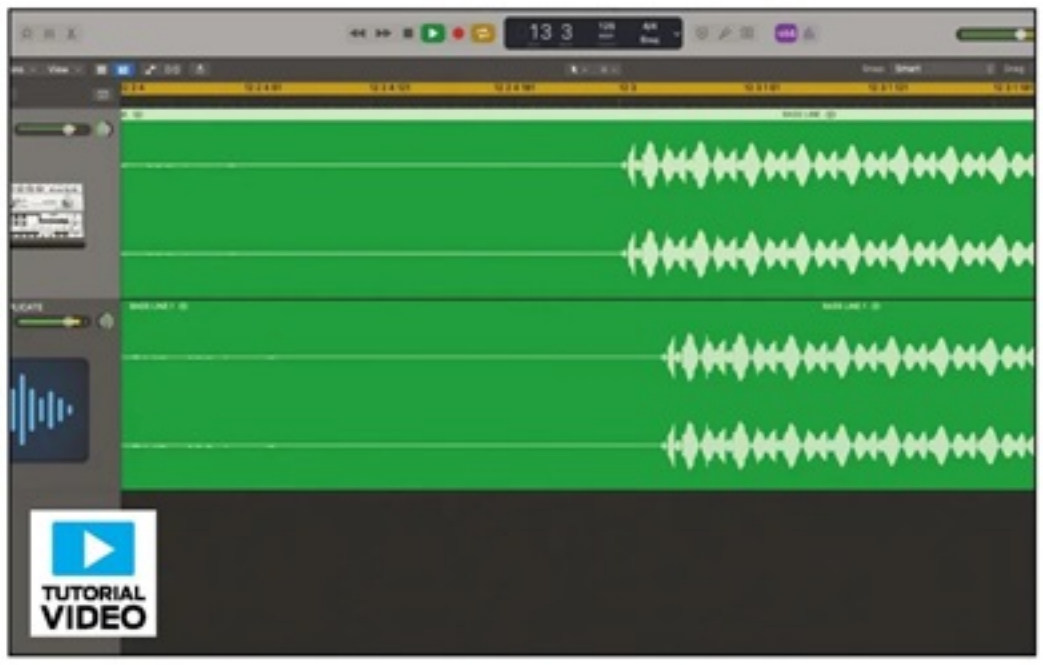
Given how handy this sort effect can be, it should come as no surprise that plugin developers have finessed the idea and bundled it in task specific plugins.

These definitely vary in complexity, and you'll even find a free one (JST & Boz Digital Sidewidener) in the **cm** Suite. I'll finish up by looking at what these can add.



## > Step by step

## 1. Ways of adding stereo width to mono sources




1 > If you want to add stereo width to a mono signal the simplest way is to split it into two, hard panning the two signals in opposite directions and then delay one side slightly. This works for delay times up to about 25ms (the Haas effect). Any longer and you begin to notice the delay.

2 > You can do this by duplicating the track and delaying one of the tracks. Or simply grab a utility delay plugin with millisecond delay capabilities. However, although this technique sounds great in stereo, when you mono it you will probably get comb filtering.

3 > Whether the comb filtering in mono is a problem is, of course, subjective but also quite dependent on the type of sound and the exact delay timing. One option is to mono the signal and adjust the delay timing to get a sound that's acceptable in mono.

4 > If we combine delay with pitch modulation then we can reduce the comb filtering created by our delay technique and this is what a chorus does. However, to make this work as a widening effect, we need to split the effected and direct signals, much like the classic stereo chorus effect, panning one to the left and one to the right.

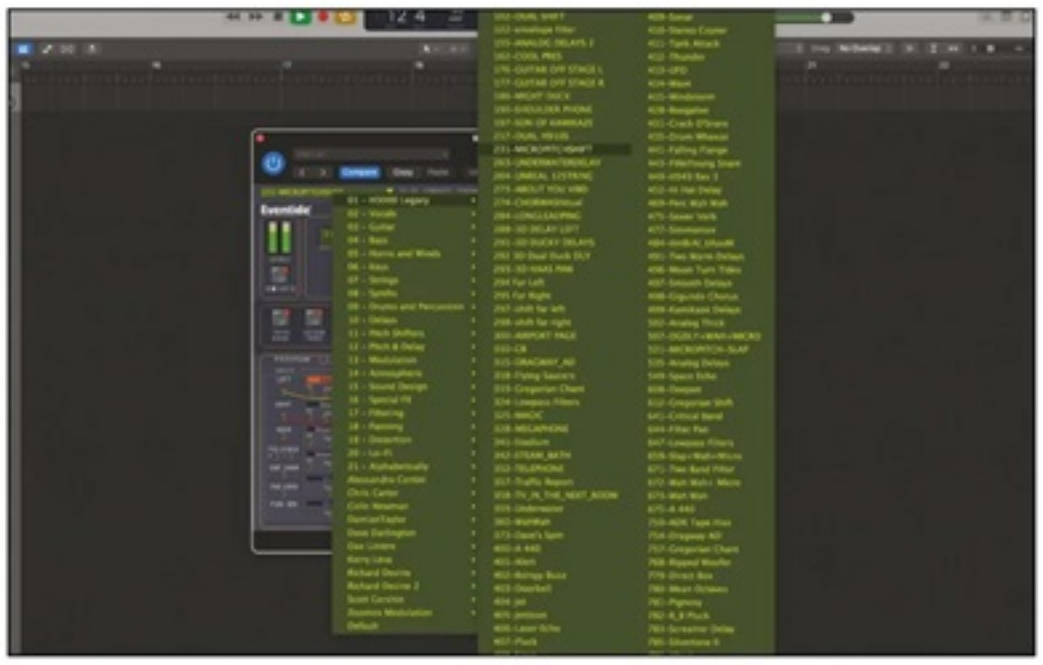
5 > You'll find that some plugins include this as an option. A plugin such as UAD's Brigade Chorus, which is based on the famous Roland CE-1, includes the Classic mode for this purpose. Alternatively, grab a chorus with dual mono capabilities, setting the mix to 100% on one side and 0% on the other.

6 > With the two signals basically hard panned to opposite sides this produces a pretty wide effect and it can sometimes sound a bit too detached. If this is the case you can simply use a stereo width plugin to make things narrower, or if you've used two separate tracks, try reducing the panning.

Download the accompanying video for this feature from here: **filesilo.co.uk/bookazines**

## > Step by step

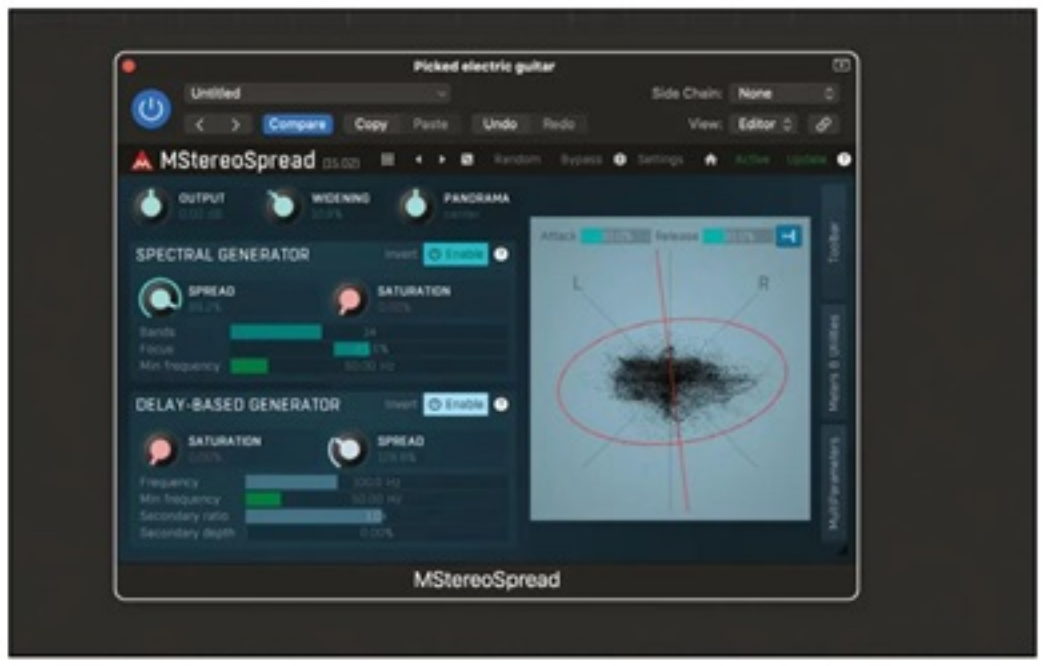
## 1. Ways of adding stereo width to mono sources *(continued)*



7 > A bit like applying delay, we can create width by shifting pitch. Typically associated with Eventide's H3000 MicroPitchShift effect, the key here is that the shift is very small (a few cents) and that it's different on the left and right. It can also be combined with a short delay like in step 1.

8 > To match the classic effect, you're looking for about +9 cents on one side and -9 cents on the other. Not all DAW stock plugins can deliver this sort of shifting in a convincing manner. Plugins such as Soundtoys Microshift and Eventide's MicroPitch are purpose built for the job, but more general plugins such as Audio Damage Discord 4 are also perfectly good.

9 > A variation on this is to modulate the pitch on one side using an LFO. This is a simplified version of another H3000 classic (Voice Doubler). Using H3000 Factory, I've augmented the existing micro pitch shift, setting the LFO to 1Hz and the shift limit to about 10 cents. Other good plugins for this include Audio Damage Discord 4 and Waves Doubler.




10 > Mono to stereo plugins vary in their complexity and effectiveness but usually attempt to retain good mono compatibility. Couple this with their turnkey simplicity and they present a highly attractive proposition when you're after a quick reliable way to add width to mono sounds.

11 > Sidewidener is a free **cm** Suite plugin with three width modes that provide progressively more decorrelation and thus wider effects. Simply adjust the Width control to taste and use the Tone parameter to focus on the desired frequencies. It's simple to use, doesn't degrade your mono signal and is, of course, free.

12 > If you have some budget, Nugen's Stereoizer 3 is one of the best wideners around, handling both mono and stereo sources. For mono sources you have two algorithms based on timing and intensity, and both include frequency limits and focusing. It's highly flexible, sounds excellent and has great mono compatibility.

PRO TIP

## BARELY THERE REVERB

As mentioned, reverb is a great way to add width but it also adds colour, and the techniques here are designed to minimise added colour. However, there is a way to use reverb without adding too much colour, and this can be great for creating subtle width. What you need is a very short reverb. Nevertheless, the success of this technique really rests on finding the right sort of reverb and the short reverb capabilities of plugins seem to vary quite a bit. What you're after is an effect that lets you reduce the size and decay time right down and also has a nice decorrelated effect, so it adds width. To check this, load the reverb as an insert and set the mix to 100% wet. Here I've loaded up one of my favourites (IK Multimedia's CSR Room) which allows you to adjust the width of the reverb signal, so you can exaggerate this effect even more. With a few key tweaks (decay time 0.05s, room size 4.0m and high cut 2kHz) I've created a lovely wide effect which I simply need to blend to taste, which is about 40% for this guitar track.

Britney Spears' *Toxic* features a perfect example of mono to stereo

### RECOMMENDED LISTENING

**ED SHEERAN**
*2STEP*
Check out the first 30 seconds of this track to hear typical mono to stereo processing in a contemporary mix.
bit.ly/CM_2step

**BRITNEY SPEARS**
*TOXIC*
A great track to reference on many levels, but focus on the acoustic guitar for a perfect example of mono to stereo.
bit.ly/CM_Toxic

# DAW MIXING

You don't need the latest and greatest plugins to get a polished-sounding mix. We show you how to get the most from standard DAW EQ and dynamics processors

**It's a common misconception among new producers that anyone making decent-sounding music must be using hundreds if not thousands of pounds' worth of high-end plugins. The reality is that many artists simply use their DAW's stock mixing effects. The producer who knows how to get the results they want from basic equalisers, compressors and gates is much more likely to make quality mixes than the producer with a ton of expensive plugins but only a shallow understanding of how to use them. In fact, having too many plugins in your arsenal can be counterproductive: those who only have access to limited tools tend to learn to use those tools far more comprehensively.**

So, rather than spending a fortune on new effects, we recommend mastering the ones you already have to hand in your DAW first. Not only will this help you to make better mixes now, but it'll also enable you to make more informed decisions as to what types of sounds and features you require if and when you do decide to upgrade.

In these walkthroughs, we'll show you how it's possible to get the most out of the most basic EQ and dynamics processors. We'll reveal how some plugins can be used for more tasks than you might imagine: for example, EQ can make it possible to boost the overall level of a signal by attenuating peaks in its frequency spectrum, a compressor can be used to control and enhance transients by turning up its attack time, and a gate can be used to tighten up percussion samples.

By following these guides you'll increase your understanding of these simple but powerful effects, and improve your ability to mould sounds into the shapes you desire. With this knowledge you'll be able to achieve a far greater variety of results than you would relying purely on presets, and you'll get better overall mixdowns as a result.

*We're using Cubase, Live and Logic but the processes can be used in most DAWs. All audio examples are at FileSilo.*

## > Step by step 1. Shaping drums with a gate in Cubase

1 > Noise gates were originally invented to reduce the overall level of noise in multitrack analogue recordings, but they're still useful in the digital age for, amongst other things, shaping drum sounds. Create an audio track in Cubase and drag **Clap.wav** onto it.

2 > This clap has a lot of reverb on it – in fact, it's so long that the sample ends before the reverb tail has finished! This clearly won't do. To fix the issue, click the Insert tab in the Inspector, then the triangular disclosure button on the first slot. Select the Gate effect from the Dynamics folder.

3 > Even at its default settings, the Gate gives us a much more natural sound, fading the tail of the reverb to silence. Let's get a bit more aggressive with it. Turn the Release level down to 11. This gives us just a short clap sound. Now it's impossible to tell there was ever a reverb on it in the first place. (Audio: **Short clap.wav**)

4 > You can make the sound even shorter by turning up the Threshold level, if you like. Let's set it to -14dB. Now we've got our shorter clap, there's nothing to stop us adding our own reverb for a different sonic character. Add REVerence after the Gate effect in the Insert strip.

5 > Click the empty patch name slot to bring up the list of presets. Select Plate At 4sec and turn REVerance's Mix level down to 15. This reverb is very intense, so it sounds pretty loud, even at this low level. (Audio: **Clap with new reverb.wav**)

6 > Add another Gate after the reverb, and turn the Release down to 61. Again, this tightens things up, giving us a more usable reverb tail. You can A/B the original and new versions of the clap by clicking the Insert strip's Bypass button. As you can hear, we've completely replaced the character of the original reverb with a different effect. (Audio: **Tightened reverb.wav**)

## > Step by step 2. Using compression to enhance transients in Ableton Live

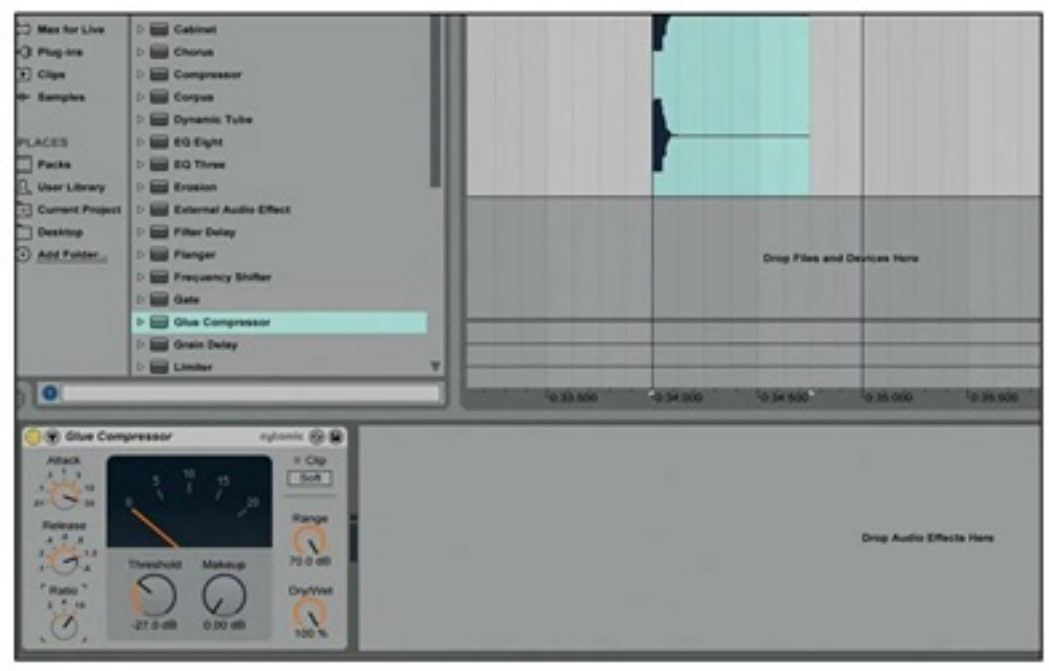
**1** > Compression isn't just for reducing dynamic range and increasing average volume level – it can also be used to enhance the transients of your drum sounds. This is useful when you have a drum sound with the right character, but that's lacking that initial hard attack. Launch Live and drag **Snare.wav** onto an audio track.

**2** > As you can see from the snare's waveform, its dynamic range is pretty low. The fat sausage shape at the start means the beat is indeed very loud, but what if you need more impact at the beginning of the sound? Add Glue Compressor to the track.

**3** > We'll use compression to lower the level of the sample, but only after the transient stage has passed. To do this, turn the Attack parameter up to 30ms. This means that the compression takes 30ms to reach its full effect after the Threshold has been exceeded.

**4** > Turn the Threshold down to -27dB. On playback, you'll hear that the sound has less perceived volume overall, but the transient at the start of the snare is now much more solid and satisfying. (Audio: **Snare attack.wav**)

**5** > The Release knob determines how long it takes the volume level to return to normal after the signal drops below the Threshold. The longer this is, the longer the compression lasts. Turn it right up to 1.2s to get as tight a sound as possible. (Audio: **Snare release.wav**)




**6** > Now turn the Ratio up to 10 to give us the most severe compression Glue Compressor is capable of. With everything set up, you can use the Threshold knob to fine-tune the balance between prominent attack and overall loudness. (Audio: **Snare ratio.wav**)

> Step by step

## 3. EQing beats with spectral analysis in Logic

**1** > When you're dealing with a sound that has extreme frequency spikes – as some drum loops and breaks do – spectral analysis can help you locate their positions. Create an audio track in Logic and drag **Funky jungle.wav** onto it. Set Logic's tempo to 140bpm.

**2** > Click the Audio FX button on the audio track in the Inspector, and select EQ»Channel EQ»Stereo. Click the Analyser button on the left-hand side of the Channel EQ's interface, and when you play the project back you'll see the audio signal represented visually behind the EQ curve.

**3** > Channel EQ's analyser isn't particularly configurable, so let's use something more suited to the task. Add Voxengo SPAN (free download from voxengo.com) after the Channel EQ, and use the horizontal bar at the bottom of its frequency display to zoom in on the high end of the frequency spectrum.

**4** > Using SPAN, we can see that the peaks occur at 9.4kHz and 12.2kHz. Return to Channel EQ and set two band EQs with Qs of 2.00 to take off 10dB at 9,400Hz and 12,200Hz. This tames the peaks, giving you more headroom to turn the channel up and make it louder in the mix. (Audio: **EQed beat.wav**)

## Four classic equaliser plugins

**DDMF IIEQPRO £27**
The commercial version of DDMF's IIEQPro includes all the great features of the CM Plugins version (such as serial and parallel routing options and 19 filter modes, including several Butterworth filters) plus more bands and a built-in spectral analyser. It's also very reasonably priced.
**ddmf.eu**

**VOXENGO GLISSEQ $69.95**
From the maker of SPAN comes this incredible equaliser that not only has the same powerful spectral analysis capabilities as SPAN, but also boasts program-dependent band response. This dynamic approach to EQ makes GlissEQ ideal for boosting highs and lows without making them sound harsh or muddy.
**voxengo.com**

**FABFILTER PRO-Q £124**
One of the best all-round EQs on the market, Pro-Q is overflowing with awesome features. With up to 24 filter bands, zero-latency and linear phase modes, intelligent solo mode for auditioning single bands, and undo/redo and A/B comparison functions, it's a real equalisation powerhouse.
**fabfilter.com**

**WAVES REDD $249**
Based on the EQ sections of Abbey Road Studios' legendary mixing consoles, Redd delivers that classic 60s sound. As well as the simple EQ controls themselves, you also get Drive and Analogue knobs for dialling in the perfect level of vintage warmth.
**waves.com**

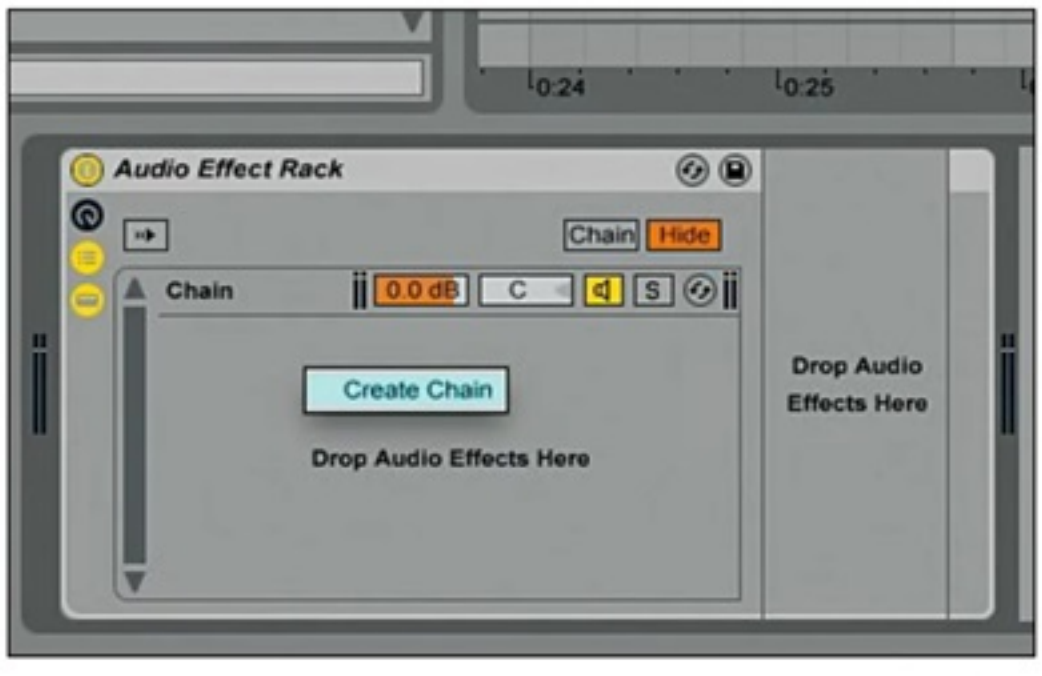
## > Step by step 4. Parallel processing in Live

**1** > Parallel processing is a technique whereby a processed (for example, compressed) signal is mixed with the unprocessed version of itself. In the case of compression, this would give us a balance of loud 'wet' signal and punchy 'dry' signal. Start by dragging **Funky jungle.wav** onto an audio track in Ableton Live.

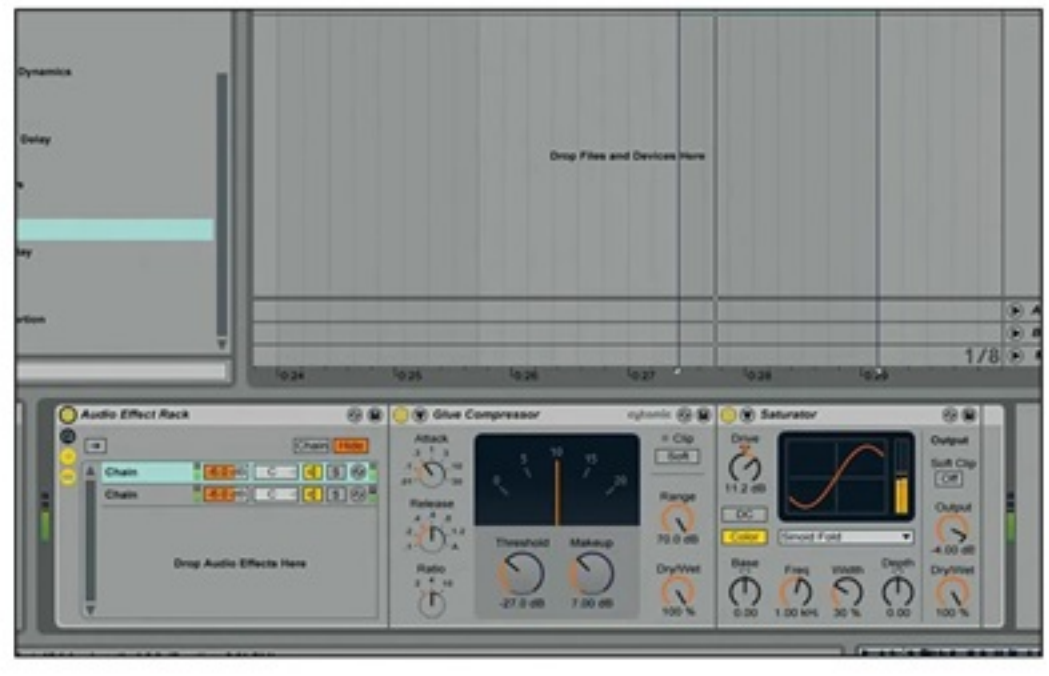
**2** > Drag Glue Compressor onto the track. The effect has a Dry/Wet knob, which means that we can mix some of the dry signal back in directly from the plugin itself. However, by setting up a parallel processing chain we can use multiple effects, which can then be mixed with the dry signal at the end.

**3** > Set the Glue Compressor's Threshold to -27dB, Makeup to 7dB, and Attack to 0.3. Add a Saturator after it, with its mode set to Sinoid Fold and its Drive turned up to 11.2dB. The processing we've added has boosted the peak level by a few dB, which we can check by bypassing the effect.

**4** > To compensate for this, turn the Saturator's Output down to -2dB. Now the signal peaks at around the same level as the unprocessed signal did, although the dynamic range has been significantly reduced, so it hits the peak level much more often. Now let's create our parallel routing. Select both effects, right-click and select Group.




**5** > This puts both effects in an Audio Effects Rack. Click the Show/Hide Chain List button and a list of chains appears in the Audio Effects Rack. Currently, there's only one chain in the Rack. To add another, right-click below it and select Create Chain.



**6** > Now we've got our 'wet' chain with effects on it, and a new 'dry' chain for the unprocessed signal. As we've got two versions of the signal, the output of the track will be very loud, so turn both chains down to -6dB to halve their volume. You can use the Chain Volume parameters to balance the signals, and toggle the Audio Effects Rack's Device Activator button to compare the parallel processed signal with the unprocessed version. (Audio: **Parallel processed beat.wav**)

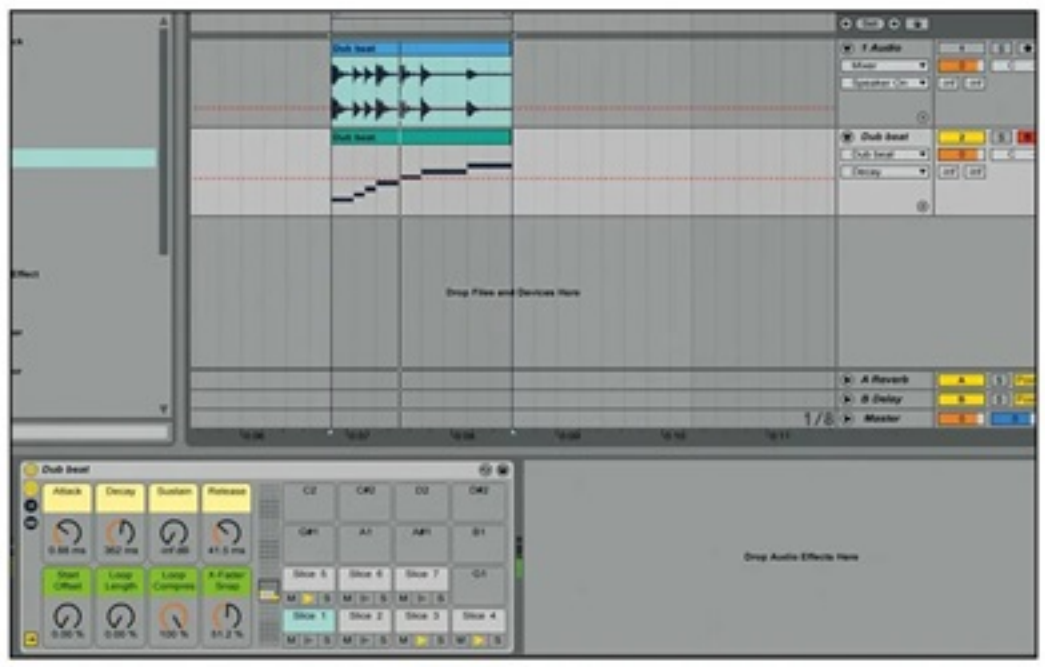
> Step by step

## 5. Tightening drum loops with Live's Slice to MIDI function

**1** > Using drum loops or breaks is a convenient way to give your beats extra rhythm and character, but if they're too loose or have too much reverb, they'll need tightening up. Doing this manually can be fiddly, so some DAWs have their own beat-slicing capabilities to make it easier. Drag **Dub beat.wav** onto an audio track in Live.

**2** > Live will automatically set the project tempo to 140bpm. Select the clip and press Cmd/Ctrl+L to set loop points around it. This loop is drenched in dubby reverb, but we can quickly get rid of that and tweak the loop in various interesting ways by slicing it to MIDI.

**3** > Right-click the clip and select Slice to New MIDI Track. A window appears, asking how you want it to slice the loop and which preset to use for the resulting instrument. The default settings of slice by Transient and Built-in slicing preset will be fine for our purposes. Click OK to slice the loop.




**4** > A new MIDI track with an Instrument Rack loaded appears below the audio track. Mute the audio track. Now we can tighten up the new version of the beat on the MIDI track by turning its Sustain down to -infdB. Turn the Decay down to 362ms for a super-tight version of the beat. (Audio: **Tight dub.wav**)

### Make your own slicing presets

Live's built-in slicing preset is convenient because it enables quick access to the ADSR envelope controls of each slice, giving you an easy way to control a beat's dynamics. The problem with it, though, is that it doesn't have macro controls linked to each slice's Transpose and Detune parameters, meaning that if you want to retune the loop you have to go through and adjust every slice manually! Clearly, this isn't a very convenient solution, but thankfully it's possible to create your own slicing presets with which you can set up eight macros as you see fit. That's enough to control the ADSR and both tuning parameters, with two left over for other functions – volume and velocity sensitivity, for example.

To make your own preset, create a new MIDI track and put a Drum Rack on it. Put a Simpler in the C1 slot, click the Show/Hide Macro Controls button, then the Map button on the device title bar to enter mapping mode. You can now assign macros by clicking a parameter, then clicking the Map button on a macro. Note that when you do this, the value on the parameter will reset to 0. When you've finished, exit map mode by clicking the Map button on the device title bar again, and set each macro to the default value you want it to have.

Finally, drag the Drum Rack into the User Library/Defaults/Slicing folder in Live's browser. This creates a preset, which will be highlighted, prompting you to enter a name for it. Type in a name, press Enter, and boom – you're done! Now, when you use Live's Slice to New MIDI Track function, you'll find the preset that you just created in the list of Slicing Presets.

You can find all of our audio for these tutorials in the Tutorial Files folder at filesilo.co.uk (in the Bookazines folder).

## > Step by step

## 6. Cutting or enhancing reverb with Logic's Enveloper

**1** > Apple's Logic Pro includes a unique and really quite powerful dynamics processing tool in the form of Enveloper. This can be used not only to tighten up the reverb on drum loops, but also to make them sound louder and boomier – ideal if you're after a big, 'live' drum sound. Start by dragging **Dub beat.wav** onto an audio track and setting the project tempo to 140bpm.

**2** > In the Inspector, click the Audio FX slot on the audio track and select Dynamics»Enveloper»Stereo. The plugin very effectively detects the transients in the loop, and enables you to control the tail of each hit using the Gain fader on the right-hand side of the interface. For starters, turn it down to -100%. (Audio: **Tight dub.wav**)

**3** > This immediately gets rid of the reverb and gives us a really tight, dead drum sound. You can get different curve shapes by using the Time knob to the left of the Gain fader. Set the Time to 60.00ms and the Gain to -74% for a less severe reverb reduction effect. (Audio: **Release curve.wav**)

**4** > Let's use the effect to enhance the reverb now. Set the Time to 2000.00ms and turn the Gain up to 100%. As you can hear, this has a really drastic effect on the sound. The boominess is so loud that it's going to clip the master, in fact, so set the Out Level to -9dB. (Audio: **Boomy reverb.wav**)

**5** > We can use Enveloper to treat the attack stage of each hit as well as the release stage. First, return the Gain fader on the right-hand side of the interface to 0%, then gradually turn the Gain fader on the left up as you play the beat back. Once you reach the 60% level you'll really start to hear each hit become more pronounced. (Audio: **Transient enhancement.wav**)

**6** > It's possible to dull the attack stage of the sound as well as enhance it. Turn the Gain fader down to 100% and it'll sound like you've turned up the attack time of each slice. This is potentially useful when you want to layer one-shot drums with a loop to give them character but the transients of the loops clash with those of the one-shots. (Audio: **Dulled loop.wav**)

## > Step by step 7. High-shelving drum loops with parallel EQ

1 > Using a pair of high-shelving EQs can be a very useful way to shape the high and/or low end of drum loops. To demonstrate this, we're going to use DDMF's IIEQPro CM, which you can find in the CM Plugins folder at FileSilo. Create a new audio track in your DAW and drag **Dub beat.wav** onto it.

2 > Insert IIEQPro CM onto the audio track. Click the Off button on the bottom row to pick an EQ type. Select HighShelf and an orange node appears at the left-hand side of the graphical display above. Drag this to the +5dB position at 2000Hz.

3 > Now click the Band 1 button and select Band 2. Set this band to HighShelf as well, and this time move the node to -7dB at 100Hz. The white line displays the overall effect of the two bands, revealing that by using two high-shelf EQs we've created a curve that we couldn't have made using just one. (Audio: **Serial high-shelves.wav**)

4 > Click the Serial button on the right to switch to IIEQPro CM's unusual Parallel mode. In this mode, each band works on a dry version of the signal rather than one that's already been processed. This affects the overall sound of the EQ, illustrated by the change in the shape of the white line. (Audio: **Parallel high-shelves.wav**)

5 > As well as using the Frequency and Gain parameters, we can also use Q to control the shape of the EQ – and IIEQPro CM's Q can get pretty extreme. For example, by tuning Band 2's Q up to 3.00 we can remove the low end of the reverb from the beat while simultaneously boosting the kick. (Audio: **Boosted Q.wav**)

6 > Finally, let's try out IIEQ Pro CM's all-pass filter mode. In serial mode this doesn't do a whole lot, but in parallel mode it has a quite dramatic effect. Select Band 3 and set it to APF. Now, when you sweep its frequency up and down, you'll see that the overall EQ warps around its position – this is ideal for experimenting with different frequency responses. **cm**

# WILLIAM ORBIT

William Orbit is the master of the mixing desk. The uber producer has worked with Madonna, U2, Blur, Robbie Williams and Britney Spears, but his career then went on hold. The joy of painting and a Pro Tools course got his mojo back and he has now released his first album in years...

**> William Orbit has produced and collaborated with some of the biggest names in pop including Madonna, U2, Blur, Pink, Robbie and Britney to name but a very few. He's won countless awards and sold millions of records, not to mention becoming a leading figure in the electronic music scene with releases including his incredible *Strange Cargo* series and his classical reinterpretation album *Pieces In A Modern Style*. But as mighty as all of these achievements have undoubtedly been, it's the parts of his life that link all of these peaks together that are perhaps more incredible.**

There were the times in the '70s where Orbit became a "seasoned squatter", and busked his way around Europe. In the '80s his first band Torch Sing were signed to Miles (Stuart's dad) Copeland's IRS Records – the first life changer. With the proceeds of that, he created Guerrilla Studios where he honed his production skills with the likes of The Cocteau Twins, Erasure, Gary Numan and, yes, Ricky Gervais. Then there was his first hit single: not an ambient electronic masterpiece, rather Harry Enfield's *Loadsamoney (Doin' Up The House)*. His single with Bassomatic, *Fascinating Rhythm*, was much cooler, as were his remixes – he reworked *Radio-Activity* by Kraftwerk while the band's legendary Ralf Hütter was in the studio with him.

This all led to Orbit's successful solo career and production work on era-defining records – Madonna's arguably greatest album, *Ray Of Light*, and Blur's *13* – and collaborations with Beck, No Doubt, Ricky Martin, Queen, All Saints and many more.

Polly Scattergood's distinctive vocal shimmers on *Colours Colliding*

After collaborating on her album in 2010, Katie Melua and Orbit join forces again on *Duende*

Orbit says he and Beth Orton, "always clicked and there are so many things I love about working with her"

But just when some would consider retirement, Orbit's life took an unexpected dark turn. He developed a serious drug habit in his early 60s, to the point of developing psychosis; during one episode he believed he would be assassinated by the Church of England and that he worked for the royal family. Luckily, a spell in rehab cleared his mind but by this time, and after a series of less well-received releases, he had lost his confidence and the will to work with anyone, nor release any new music.

Which makes last year's album, *The Painter*, all the more extraordinary. And it's a real return to form – there are some stunning tracks on the album, with collaborations with Beth Orton, Georgia, Katie Melua and Polly Scattergood – but how did this seismic change in fortunes happen? The answers are typically Orbit: becoming a painter inspired him and gave the album its name, but it was a beginner's course in Pro Tools which really reopened the production floodgates. "It was inspiring," he says. "Once I got started, it started to flow and then it was a torrent and it felt like I was in 1997. It was a release, you know?"

**cm: The album *The Painter* has some extraordinary vocal talent on it. Was it easy to get them all involved?**
**WO:** "Sometimes it's just a matter of getting the manager's email address and saying, 'how about it?'. And in this case it worked. And it was wonderful because I haven't been doing this for a long time, and coming back it wasn't at all difficult. It almost felt like it fell into my hands. And the same with the singers who all gave me their most passionate and wonderful vocals. I couldn't have asked for better."

**cm: Did you record the vocalists remotely or did they come into your studio?**
**WO:** "A couple of times I already had the vocal and then built a track around it, the Beth ones for instance. Hers were recorded at the beginning of the '90s, a good 30 years ago. With Katie we did it in a Las Vegas. The rest were all long distance because of Covid. Natalie [Walker] was in Colorado, Lido [Pimienta], although she is Colombian is in Canada, and various other people were singing it in. Normally they would hear a track that I had started and then send me their vocal and that would be the one that I used. Technically speaking it's a bit basic in some respects as some people recorded theirs in a kitchen but it doesn't matter."

**cm: Did you ask Madonna to appear?**
**WO:** "No I didn't, both of us are doing our thing so it would have meant a lot of manager's talk. She's firmly attached to a project she's currently on which is going well and I'm firmly attached to this, so we didn't but I wouldn't rule it out in the future. I did unearth a track called *Cuidado Con Mi Corazon*, a version of a track that she and I did for Ricky Martin that has this kind of tango beat. I'd done a reworking of it just with

"Once I got started, it started to flow and then it was a torrent and it felt like I was in 1997. It was a release, you know?"

some of her Spanish vocals. We did think about putting that on the album but the thing about albums is often what you leave off, you know? I did discover a lot of unreleased material I did with her and maybe it will see the light of day one day."

**cm: The album came after quite a few years of not making music. What happened?**
**WO:** "I couldn't face music, or even look at my equipment or open up a laptop. I had got into the songwriting for hire game and found that didn't really work so well for me. I wasn't getting a response. I felt like I was getting a wall of indifference; nobody gave a toss. I lost confidence in my own abilities to be able to make something special and I had the perfect storm of personal issues building over this period so music was the last thing on my mind."

**cm: And the album is called *The Painter* as you got into painting instead of music...**
**WO:** "Yes after I pretty much stopped a few years ago I started painting which I loved. Although I know I'm not a painter, I'm a musician, I got into painting in the same way I first got into music. I did tons of stuff in a very short space of time, very inspired, but music got pushed to the back."

**cm: But we hear that a Pro Tools beginner's course helped bring it back – so why 'beginner's' after so many years of using it?**
**WO:** "I wanted to know about some of the features that I've always worked around. I'm pretty fly when it comes to these apps but was like 'what does this one do that has been staring me in the face for yonks?'. Or 'how can I do this easier? Isn't there a shortcut?' I've seen people in studios and they do everything with shortcuts so I thought 'I really need to make the effort'.

"It's never too late to learn. And that is also key to inspiration, because you find a new thing and think, 'I must apply that to something'"

"And then I realised how much I *didn't* know. As you race along and never read the manual, you leave stuff behind. That's great because you find out where you want to get to. However, I took this very good course online with a syllabus and a good teacher. I didn't carry it on for that long but it was marvellous."

**cm: So unleashing some of Pro Tools' power got you back into music production?**
**WO:** "Yes it did, very much so. Something new and learnt was very much part of it. Every day you learn a new trick and that it's never too late to learn. And that is also key to inspiration, because you find a new thing and you think, 'I must apply that to something,' and right away you're taking this new found enablement and turning it into a musical substance."

**cm: You had been through a struggle with drugs a few years back so is it fair to say that**

# Painting by notes

## How does your love of painting translate into the album's sound?

"I wanted it to be a bit closer together in style than I usually do. I wanted it to be very vivid because I was so busy with colours – they obsessed me. I don't see colours as sound or sound as colours – synesthesia or whatever it is called. But at the same time, the process of sculpting and moulding, dissolving and finding context... All those things with painting and then doing the same with sonics, there is a parallel. So I wanted it to be that I could paint my way through this album and so I did.

"There's an image for each track and it's been a joy to do. And it was just producing tracks that would make me feel happy when doing them. It's that state one gets into, that kind of reverie. That's all it needs. If I'm in that state I can listen to each molecule going by, and each bar sounds like it should be linked up forever. Then I'm a happy camper. There was no commercial or contemporary 'what's going on in pop' kind of thing whatsoever."

"The process of sculpting and moulding, dissolving... All those things with painting and doing the same with sonics, there is a parallel"

Photo © WENN Rights Ltd / Alamy Stock Photo

# Orbiting artists

**You have obviously worked with many artists. Any favourites?**

"The answer to that should be 'a lot of them'. I loved working with Bono. What a joy to be working with a singer like that. Also Beth Orton. We've always clicked and there are so many things I love about working with her. I've loved working with Madonna: moments of just musical magic. Working with Blur for the first time, they had this tiny, little studio that Damon [Albarn] had at the time. It was at the peak of a heatwave and we were all crammed into this tiny room, and they all plugged in their guitars. Dave [Rowntree] was in the mix room drumming. Graham [Coxon] and Alex [James] had their guitars kicking off and Damon picked up the mic and it was like, 'oh I love bands, I love that collective sound, oh yes'. There was something that was just gorgeous. The session I did with Beck, we wrote a tune for Pink, and that was brilliant, we had such a laugh. I've had some very funny sessions like that where everyone's cracking up.

"So I've had great times and it's hard to pick one. There are fun, crazy sessions where everyone's wild and probably drinking too much and something comes out of it, and there are ones that are a struggle and suddenly there's a breakthrough. I enjoy them all."

**alongside painting, Pro Tools saved you from drugs?!**

**WO:** "Goodness me I have to de-compact that question! Pro Tools is an immersive application that I particularly love. My favourite apps are Pro Tools, Photoshop and Final Cut Pro. They are so immersive: creative worlds that you make your own. So creativity saved me from drugs. Because I got into drugs belatedly in my life. My first druggy period was when I was 14 and I stopped when I was 17. And then my next one was when I was about 61! Pro Tools is certainly a lot better than drugs, that's for sure."

## *The Painter*'s tools

**cm: The album certainly has many trademark William Orbit sounds. Did you use a lot of old gear on it?**

**WO:** "Yes there are a lot of old sounds. I did use some old gear – I have a lockup full of synths but I'd actually lent them all out to people as they were not getting used. What happened with this was that I had used multitrack tape [to record] from '83 to '97 when, with the Madonna album, I went over to America and went 'Pro Tools'. So over this 14 or 15-year period I had amassed about 110 2-inch multitracks packed full of recordings so I had them all digitised. Some you had to put in the oven to bake them or scrape the oxide off as they were getting a bit corroded, but I got it all onboard, a vast archive of material that's never been released, all recorded in analogue through compressors. And I did use some of it as sound sources [for the album] even if it was sometimes using it almost like a synth patch as a starting point.

"So the album took 15 months to make or 15 years. Both, if you like! And it was all done 'in the box' but before 'the box' was invented!"

**cm: So do you have your entire studio on a laptop now?**

**WO:** "Yes and no. Yes it is but I always come out at the drop of a hat. But yes I do everything in a MacBook Pro."

**cm: What are your favourite plugins?**

**WO:** "The firm favourites forever are the really prosaic ones like EQ, compression and filters. I use a Focusrite three-band EQ, the Red one, pretty much for everything. It's always been my way. I'll find one EQ I really like and I'll use it for everything. Ditto reverbs. These days I tend to use one particular Ativerb, one outboard PCM70 for everything. People think I have an array of reverbs but it's really just one or two. It is the sound that's going in that determines the sound coming out, after all.

"Really I just have a set of really handy ones [plugins] that are usable to shape and sculpt sound. They are like a palette knife, or a chisel or scissors. They are tools that are reliable and haven't really changed their design much."

**cm: What about favourite instrument plugins?**

**WO:** "Some of the sound-crafting ones are awesome. I do like [Native Instruments] Massive, but they have to fix this thing where notes just hang. It's a real problem, they just hang and it's not a MIDI thing because you can't just set all MIDI to off. I like a library like the next man. If I haven't got a real string section I use libraries joyfully. Kontakt is awesome and I love Melodyne as a remedial tool as well as a creative tool. I really like Serato – obviously huge in the

# Best classic hardware synths?

**cm: What classic hardware synthesisers have been your favourites over the years?**

"The Korg MS-20 has been a favourite: wildly unpredictable and if you get a good sound you need to get it down onto tape asap because it won't be the same in a minute. The Roland Juno-106. I've been into Junos since the Juno-6 back in about 1980. I remember going to Tokyo to go to their factory at the time. Then I had the Juno-60, then the Juno-106. It's my number one keyboard on stage. *Ray Of Light*, all the *Strange Cargo* records, half of it is the Juno-106. It's just a fantastic sonic, a simple machine, easy to just grab faders and use them in a performative way. I love the JD-800 for its sonics too but it is not as intuitive. It's big and takes up a lot of room but it's got such a complicated waveform to it that it is so rewarding to use once you find out how to program it.

"I've always liked Roland stuff and old analogue synthesisers. I like the ones with the traditional ADSR on it, and you've got the usual VCO controls. It's like on a mixer, any DAW mixer or one on an app – you'd expect it to have any basic controls like pan, mute, solo. Nothing is ever going to change that, whether it's on a phone or an old desk. These things are fundamental to how music gets recorded. It's never going to change. And I think it's the same when it comes to synths. There are certain things to control and sculpt sound and you have to have easy access to them. You don't want to go through pages and pages of stuff. So if a synth has the basic controls, whether it is virtual or real, then I like it."

DJ world – the pitch and time [tools], the professional ones. It is an audio suite plugin I use all the time. I use it for just lengthening notes, performing things in time, for taking tracks of different pitch and fitting things in – a host of things, it is always open.

"A really good tool is fabulous for a little invisible mending but you can also use it to rock out on some wild areas – that is the best kind of plugin."

**cm: Would you like to get back to producing other artists again?**

**WO:** "Yes. I'm moving to the north of Italy shortly, to Venice, so not hard to get to if people want to collaborate. In fact it's very nice: a few days in Venice and I'll put you up in my place or you find a nice place to stay and you get to my house by canal! So I would like to do some fun stuff, writing and working with artists again. I feel confident I've got something to offer. Sometimes working with artists they are so perfect as they are that I think, 'can I just stay a fan boy?' or, 'there's nothing I can contribute'. But there are a lot of artists where I think I could do something really cool if they are up for a bit of musical adventure. I'm ready for that. I'd love to do something with Shakira, she's my favourite. I think she's a genius actually, in terms of everything she does. I really mean that."

**cm: What else do you have planned with the latest album?**

**WO:** "Now there are a lot of things you need to do – you can't just put it up there on the internet. So there will be an Atmos version and as I mentioned, all of the tracks will have a painting to go with them and even an animated version. I'm planning an art exhibition as well. My canvases are very large so I have to stick them in a truck and drive for 24 hours to Venice and put them up in a space I have there. And collaborating with some great artists and singers, yeah I love that. Back to that again. There's nothing I like more than to work with an artist to help make their vocal sound like heaven so if you want me to help you, I'm ready to go. Here I am everybody, I'm at your service!" **cm**

## HEAR MORE

*Colours Colliding* (with Polly Scattergood)
**bit.ly/3tREBOh**

*Water From A Vine Leaf* (with Beth Orton)
**bit.ly/3Od3Q5G**

williamorbit.com

soundcloud.com/williamorbit

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Download the accompanying video for this feature from here:
**filesilo.co.uk/bookazines**

# Equalising during the mix

## This time we give you the lowdown on some basic EQ techniques that can help get things moving with your mix

Jon Musgrave

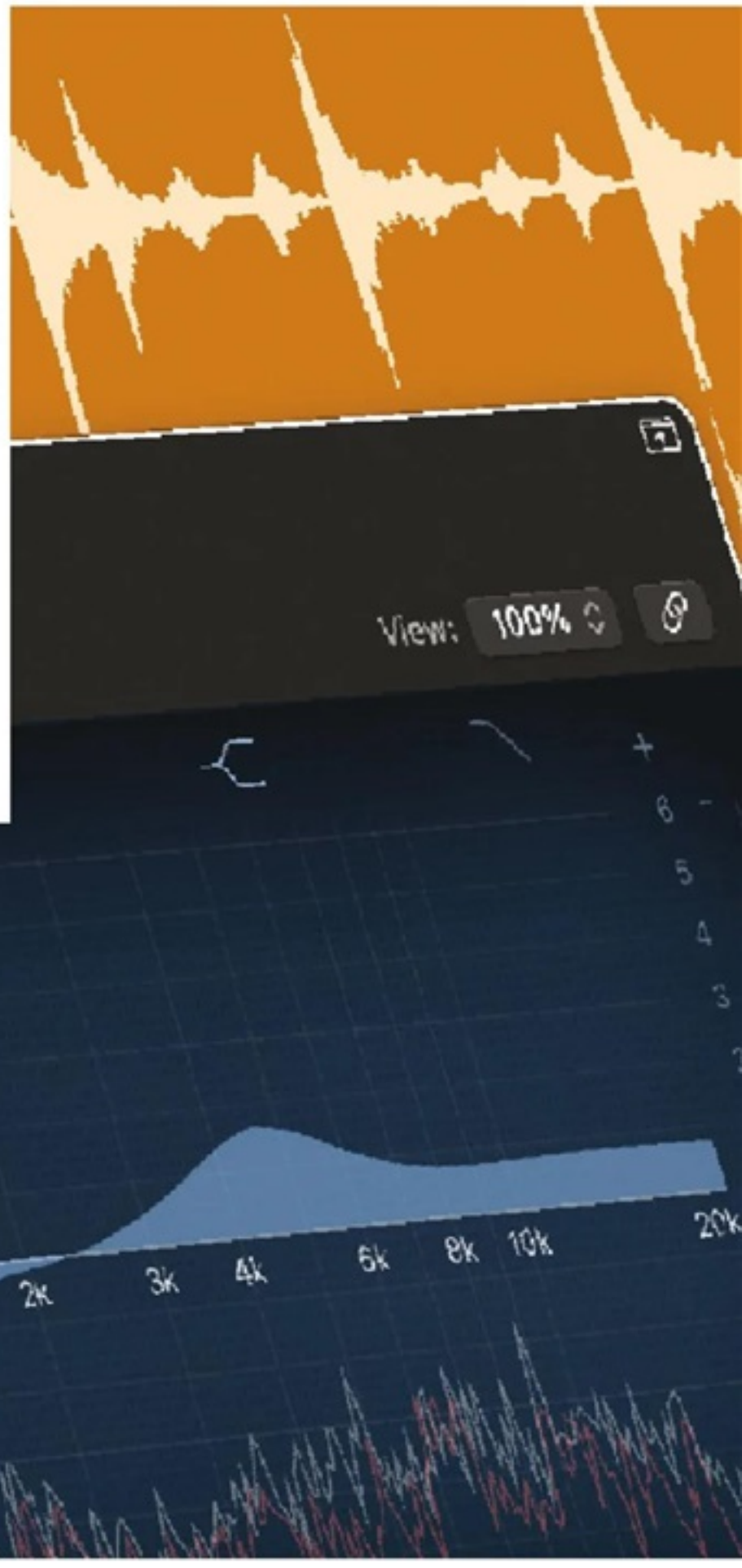
**Jon is a London-based producer, mixer, composer and club remixer with a diverse CV that spans dance, pop, rock and music for media. He's also a long term contributor to *Computer Music*. Whether working alone or collaborating he usually handles final mixdowns, which is why we've got him to share some pearls of mixing wisdom.**

**> Equalisation (EQ) is one of the most important mixing tools, yet also one that's quite easy to get wrong. But there are a few simple methods that can really improve your chances of success.**

Let's assume you already understand that an EQ includes various different shaped filters that allow you to boost and cut selected frequencies or frequency ranges. You'll typically find a pretty flexible EQ pre-assigned to all channels in your DAW and that in itself is a big invitation to get EQing. Even so, there is no requirement to EQ everything, and many would argue that you should use the minimum amount of EQ you can.

As ever in music production, there is no 'one size fits all' approach. Sometimes raw recorded sounds such as a multi-miced drum kit benefit from quite extensive EQ, whereas other raw sounds such as a solo violin need little or no EQ. Meanwhile, pre processed samples may be production ready and sound great on their own but still need specific frequencies cutting or boosting.

Whatever the target, we usually use EQ in a corrective or enhancing capacity, and at the mixing stage we combine it with level balancing to help everything fit together. In this tutorial, I'll be looking at two core techniques – sweeping and filtering – that can help you focus your EQing by finding the frequencies you actually want to process. Read on and we'll get started.

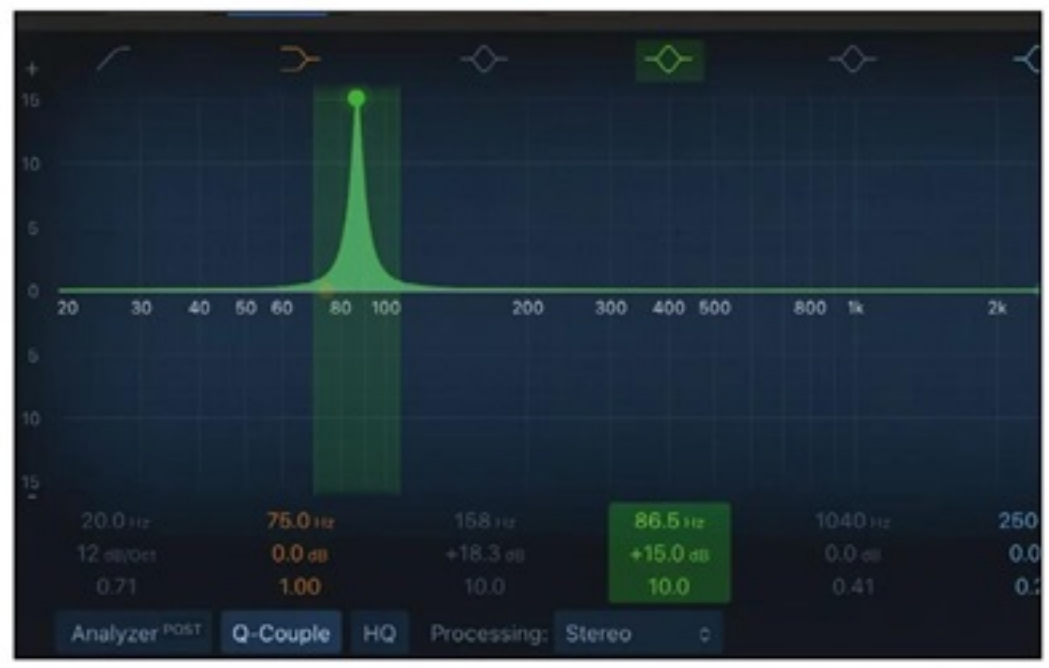


## > Step by step 1. Sweeping and filtering

1 > Even though we have plenty of visual tools to analyse signals and pinpoint certain frequencies (see Pro Tip on p49), it's important to learn how to hear these frequencies, and one of the best techniques for this is boost and sweep. This allows us to zone in on specific frequencies easily and quickly.

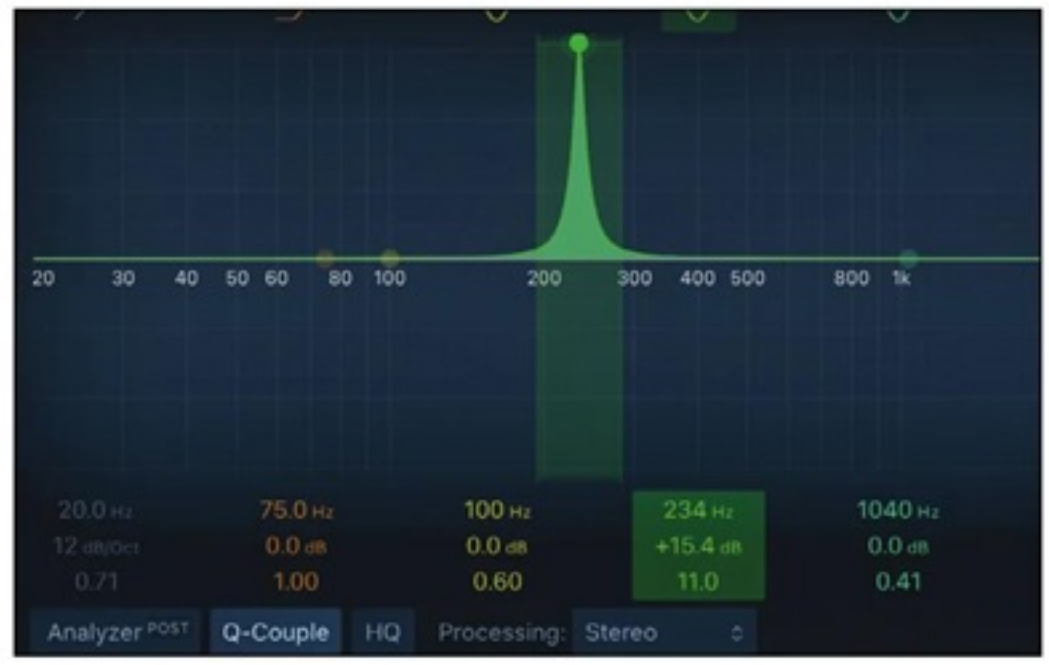
2 > The technique requires a parametric EQ with a peak shape band and an adjustable Q. It relies on the fact that we find it much easier to distinguish a narrow band of boosted frequencies and we can use this phenomenon to pinpoint areas and frequencies we want to boost or cut.

3 > We've inserted a suitable EQ plugin across our bass part, activating a parametric band and set the bandwidth quite narrow (ours is set to 10). Bear in mind some EQs allow very narrow bandwidth settings and these may be too narrow for the technique. Now add plenty of gain – here we've boosted by about 15dB.




4 > Our EQ includes a visual representation of the curve which clearly shows the boosted peak. We can easily see its width and the potential impact on the centre frequency and surrounding frequencies. Now by adjusting or 'sweeping' the frequency as we listen to the part we can zone in on its significant frequencies.

5 > You can do this quite quickly initially, and if you find that it's hard to pick out important frequencies, try widening the bandwidth until you find roughly the right frequencies. Once you have the right zone, narrow the bandwidth again to really focus on a specific area.



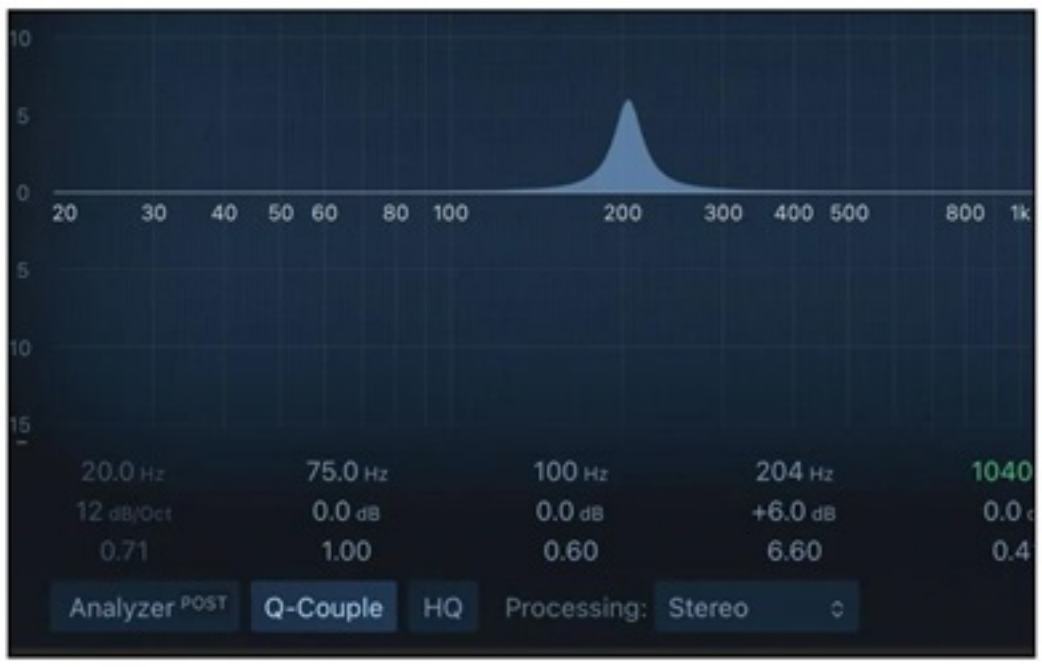

6 > Listening to our bass part, this has well defined notes and should be quite easy to process. Nevertheless, one high note in particular sticks out too much. By boosting and sweeping we should be able to enhance the relevant area, zone in on the frequency and use a narrow filter to notch it out.

Download the accompanying video and the MIDI/audio files at **filesilo.co.uk/bookazines**

## > Step by step 1. Sweeping and filtering *(continued)*

**7** > It sounds like the one problem note we spotted has a quite dominant frequency between 200Hz and 250Hz and by more precise boosting with a narrow bandwidth we've settled on 230Hz. In this instance we've decided to apply a -9dB cut at this frequency with a broader bandwidth (Q is 5.00).

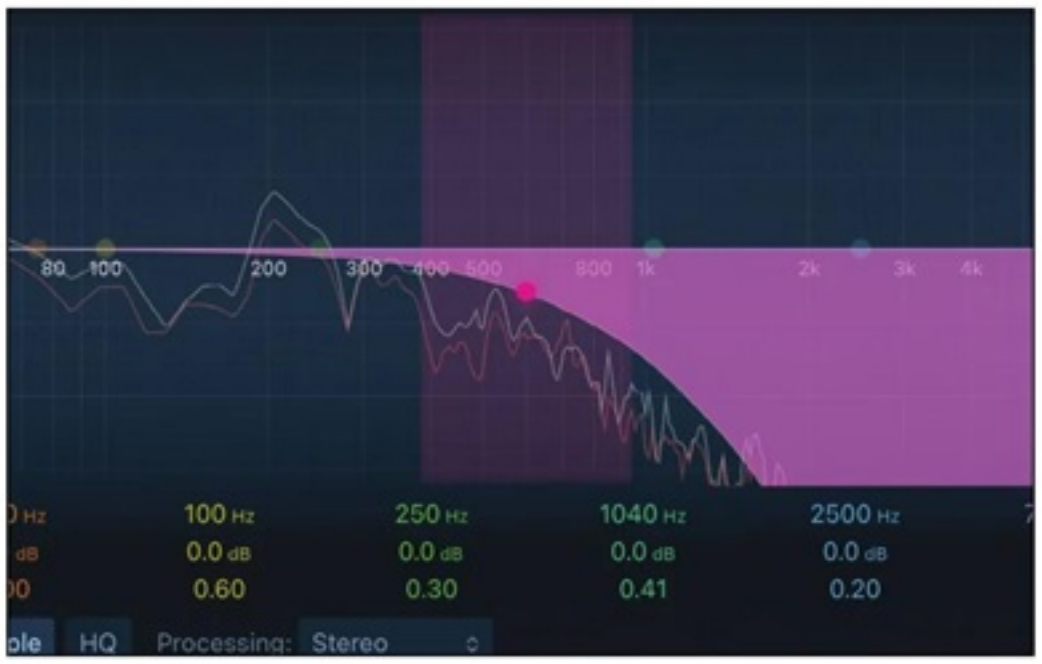
**8** > Next we have a ringy synth part with a bunch of different gnarly harmonics. By sweeping the mid to high frequencies the unpleasantness is at its worst in the 2kHz to 3kHz area. Rather than try and pick out specific harmonics, we've applied a broad cut of a few dB, leaving the higher and lower frequencies as they were.




**9** > Now on our blended drum stem we've swept the low frequencies and found the body of the snare at about 205Hz and the snare knock at around 338Hz. On this occasion we're going to boost just the body and have set a wider more natural shape peak (Q is about 7) with about 6dB boost.

**10** > The sweeping technique uses an exaggerated boost to help us focus our EQ. In contrast the filtering technique relies on temporarily removing frequencies so you can better hear a specific frequency range. You'll need regular low pass and high pass filters which you can use individually or together.

**11** > The technique is particularly effective when you're EQing more complex sounds or pre blended stems or submixes, like our blended beats stem. Here we've activated a low pass filter with a medium slope (24dB/octave) and low Q (about 0.5) so the filter shouldn't add any misleading resonance.




**12** > By lowering the cut off frequency we're not distracted by the mid and upper frequencies. With the cutoff at about 600Hz we can hear the body frequencies for the kick, snare, congas and cowbell and even start EQing them. Once you're happy, simply bypass or remove the low pass filter.

PRO TIP

## USING THE ANALYSER

One of the greatest benefits of mixing in a DAW is the plethora of available analysis tools, and EQs with their own in-built frequency spectrum analysers are a perfect example. At their best, these will show you a graphic representation of your EQ curve superimposed on a real time analysis curve (a good example is the EaReckon CM EQUA-87 equaliser in the CM Suite). This combo not only helps you EQ but also provides a great way to learn and understand the frequency make up of different sounds. If you don't have a suitable standalone analyser option in your DAW, why not try our CM Suite plugin FreqAnalyst CM from Blue Cat.

Of course, the visual feedback is only useful if we can interpret it properly. On the horizontal axis you'll find frequency, with amplitude on the vertical axis. The frequency scale is logarithmic, and each doubling of frequency, so each octave, is evenly spaced out. This puts 500Hz roughly in the middle, and means we'll often be viewing fundamental frequencies to the left hand side and higher harmonics to the right. Lower frequencies generate more obvious curves, and here you should be able to distinguish individual notes in a bass part.

Trance producer Armin Van Buuren understands the importance of EQ when mixing

### RECOMMENDED LISTENING

**DRAKE**
*GOD'S PLAN*
When you've got multiple 808s to handle, pinpointing specific frequencies can be vital
**bit.ly/cm_gods_plan**

**ARMIN VAN BUUREN**
*ANOTHER YOU (FEAT. MR. PROBZ)*
EQ is a key tool when mixing the many competing elements and layers in a trance track
**bit.ly/cm_another_you**

# MIXING WITH AUTOMATION

Take your productions to the next level by manipulating levels, mutes/solos, panning and plugin parameters like a pro

**> Back in the pre-digital days, mixes often required more than one pair of hands to make adjustments in real time as the song played. In addition to volume changes, it was often necessary to turn effects on and off, mute tracks at specific spots, move the master bus volume for a fade-out, and more. Remembering all the moves for a song wasn't easy, and it was often necessary to run the mix multiple times to get it right.**

Today's DAWs offer mixing tools that are light years beyond that. You can precisely automate anything that's happening in the mixer and effects plugins. Beyond just balancing the levels of various tracks, you can use automation to add dynamics to a boring part, mute and unmute tracks to create breakdowns, automate effects to produce stunning sound design, and more.

Further to all that, the process of automating parameters doesn't have to be exclusively done in real time by moving faders and knobs. You can create and edit automation using breakpoint editing, so draw in and edit changes, which provides you with pinpoint control over everything in your mix. The possibilities can be as vast as your imagination allows.

Here we have a variety of automation tutorials in different DAWs, old and new, the principles of which can be applied to whichever DAW you mix with.

## > Step by step

## 1. Your DAW's automation modes

1 > DAWs typically offer three basic modes – **Touch**, **Latch** and **Write** – for writing real-time automation data (the names may vary). A fourth, called **Read**, is a safe mode for playback. **Touch** mode lets you make a change with the fader, knob or button but returns to where it started when you release it. It's handy for quick tweaks.

2 > If you're in **Latch** mode, after you release a fader, knob or button, the automation stays at the new value until you move it again, rather than reverting to the original value like in **Touch** mode. For situations where you want the automation to stay at the new value (at least for a while), **Latch** mode is quite useful.

3 > **Write** (aka 'overwrite') mode should only be used for replacing existing automation. It erases previously written data as the playhead passes over it. After using any automation mode, but particularly Write mode, set the track's automation back to **Read** mode to avoid accidental overwriting of data.

## Automate dynamics to add emotion

**Dynamics – the changes in level and intensity of musical parts, whether vocal or instrumental – are key to adding excitement and emotion to your music over time. A static song that stays at the same level, 'density' and intensity throughout is likely to have less impact than one that builds into choruses, drops down for verses and breakdowns, and generally changes over time.**

**But because many musicians don't think enough about dynamics in their arrangements or their playing, and because programmed parts tend to lack dynamics, you may be faced with a mix situation in which the only way to create sufficient excitement is to alter those dynamics with automation.**

**You can globally change levels by automating the volume of the master bus. The changes should be subtle, as they would be in a dynamic arrangement played by musicians. A couple of dB up or down can make all the difference. You can also automate equally subtle EQ changes to add or subtract intensity, although these are usually better done on individual tracks. For example, adding a little brightness to the lead vocal on the chorus or bridge can bring it forward in the mix, thus adding intensity.**

### > Step by step 2. Basic fader automation

1 > View the track lane for the channel you want to automate and select volume automation. In most DAWs you'll see an automation curve (or 'breakpoint line') at **OdB**. Set mode to **Touch** for brief adjustments or **Latch** for long ones. It's now ready to capture fader moves.

2 > Hit play and move the fader as needed during playback. Do short passes or run through the whole track. If you make a mistake, either hit stop and undo it and try again, or keep going and fix it afterwards with breakpoint editing, to graphically edit the automation by drawing the curves.

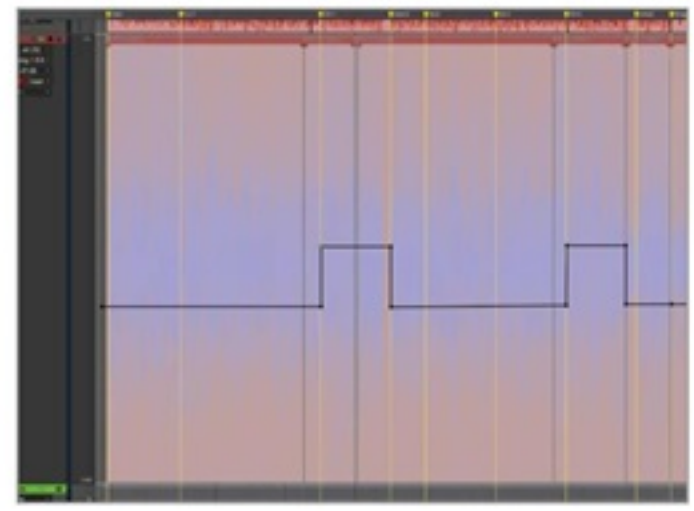
3 > When done writing the automation, set mode to **Read,** which is a safe mode for playback and won't overwrite automation levels. Hit play and listen to find the spots you want to fix or edit. The visual overlay of the automation curve on the waveform will be helpful for seeing where you need to make adjustments.

4 > The same procedure as with faders can be used for automating knobs. So, you could automate a reverb send so that reverb level increases during a particular song section; or change the amount of distortion on a guitar track by automating the gain control in a virtual amp, which we'll explore in the tutorial below.

### > Step by step 3. Automating a virtual amp's gain settings

1 > Here's an example of automating an effects parameter. Here, it's virtual-amp gain, but could be any parameter on any plugin. In most DAWs, you can start by setting the track to touch or latch mode, hitting play, and briefly moving the gain control. Now that parameter should be available for editing.

2 > Put the track's automation mode back to Read. Click the bottom edge of your track, then drag it down to to enlarge the channel's height as much as you can. The more screen real-estate available, the more precisely you'll be able to draw in your automation.



3 > Use your straight-line or freehand drawing tool to draw in gain increases where you want the more heavily distorted sounds. Typically, if you click on one of the automation nodes (the dots on the breakpoint line), you'll see a readout of its value somewhere on the DAW's GUI, which is helpful for making changes.

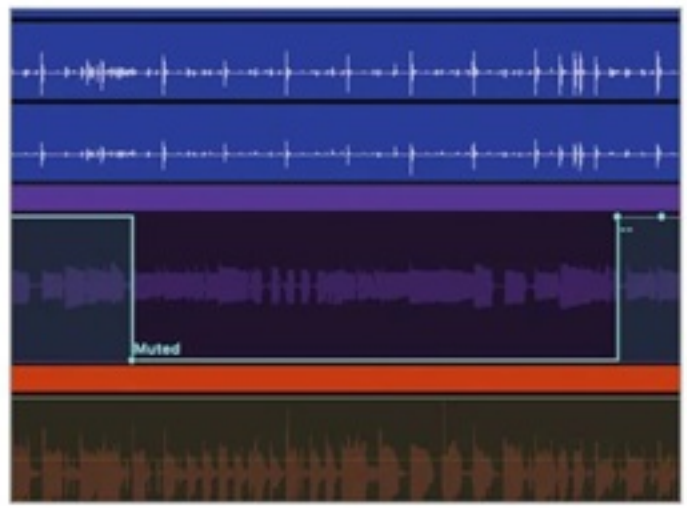
## > Step by step 4. Accentuating drum fills with volume automation

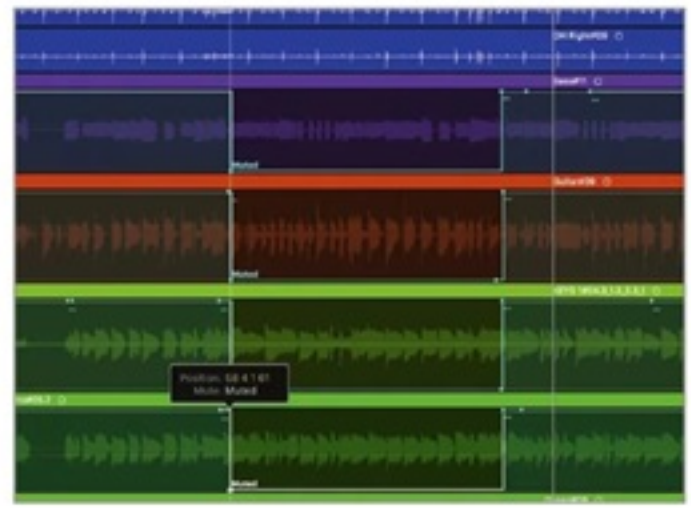
1 > If you're looking to add some energy to a song, it's often helpful to push the drum fills up a little louder than they were recorded. If you're using multitrack drums, create a sub-bus to automate the whole kit. If it's a stereo drum track, you can automate its volume directly.

2 > With the drum bus or drum track's automation mode set to **Touch**, push up its volume fader a couple of dB just before the beginning of each drum fill, and release at the end of the fill. Repeat for all the fills in the song.

3 > Use the graphic editing features to tweak the changes you made so that boosts start and end in the correct spots and at the desired level. Alternatively, create changes like this using **Touch** mode to automate them in real time.

## > Step by step 5. Creating a breakdown by automating mutes



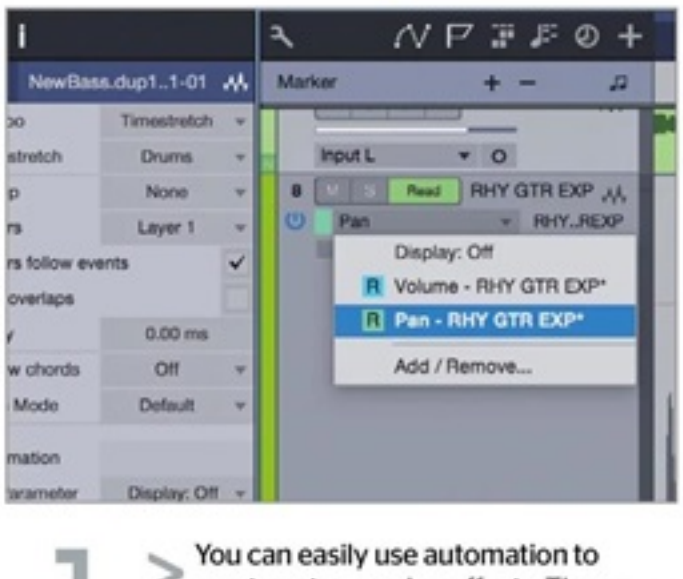



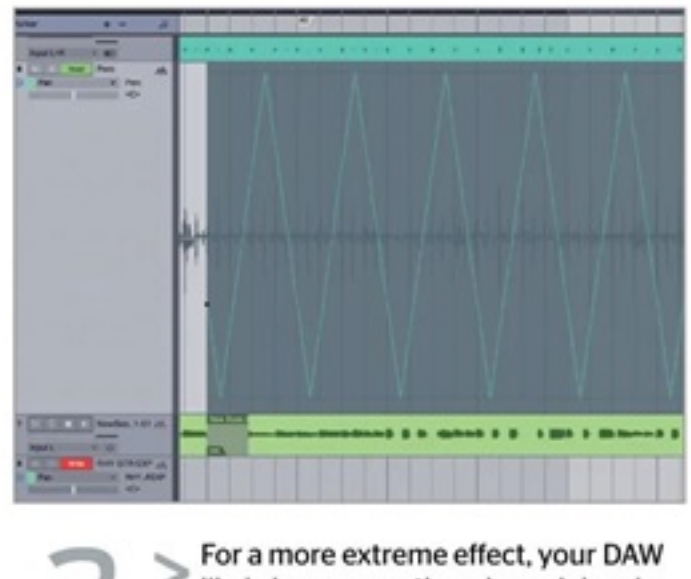

1 > To add interest in an arrangement, put a breakdown in for the latter part of the song. Automating the mute buttons on individual tracks makes it easy. Once you've decided which tracks to mute, set one to **Latch** mode and hit play, starting a couple of bars before the section.

2 > Mute the first track and release the button at the end of the breakdown section. Then choose to view mute data in the track lane, and adjust the boundaries graphically if needs be. If you want the data to start or end off the rhythmic grid, you might need to turn your grid snap off, depending on your DAW.

3 > View the mute data in the other tracks you're turning off in the breakdown and use your DAW's editing tools to draw in mutes at the same point on those tracks. Hit **Play** and your breakdown should start and end precisely where you want it to.

## > Step by step 6. Creating precise auto-pan effects with automation





1 > You can easily use automation to create auto-panning effects. These can be intense, understated, or anywhere in between. First, set the track in question to show pan automation (shown here in Studio One).

2 > An efficient way to create left-to-right movement is to use your DAW's line-drawing tool to draw in data. If you want the pan to go fully side-to-side, draw the lines to the top and bottom of the track lane. In many DAWs, if you turn on snap to grid, it will help you draw a more even line.

3 > For a more extreme effect, your DAW likely has some other shaped drawing tools, which can draw data in shapes like triangle, saw and square shapes. Used with snap-to-grid, these can yield interesting results. Use the same method for other effects such as filter sweeps.

## Be careful with automation levels

**Editing automation data can be tricky. Say you want to automate a track's volume so that it dips by 3dB for a couple of beats, and then goes back to where it was. One way to achieve that would be to use your DAW's line-drawing tool to draw in the change.**

**Once done, you may notice that there was a node created at the spot where the attenuation began, another where it started to go back up, and one at the level you returned to. The nodes essentially function as 'hinge points' at spots where the line is changing in value.**

**Here's the tricky part: if you accidentally brought the automation curve up 5dB after bringing it down 3dB, then your volume would start at 5dB and gradually return to 3dB at the next spot there's a node. What's more, if that node is many bars away, you might not even notice that your track was doing a slow decrescendo. So, whenever making those type of changes, always check the level at the start and end of the edit. It's also useful to periodically zoom out to where you can see the whole song, and check you don't have any unintended changes as a result of your edits.**

## > Step by step 7. Automating delay throws

1 > A delay throw is a popular effect that delays particular words or notes. You'd typically do it at the end of a line so there's space for the repeats to be heard. Here's how to do it using automation. First, insert a delay on an aux return track.

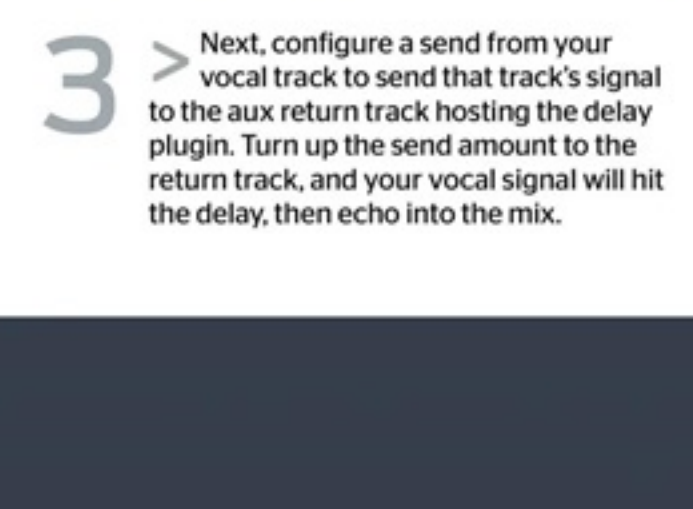
2 > Get the delay set to the time you want (1/4 notes and 1/8 notes often work well), then set **Feedback** high enough to give you the number of repeats you want. As the delay is on an aux, make sure the plugin's wet/dry level is set to **100%** wet.

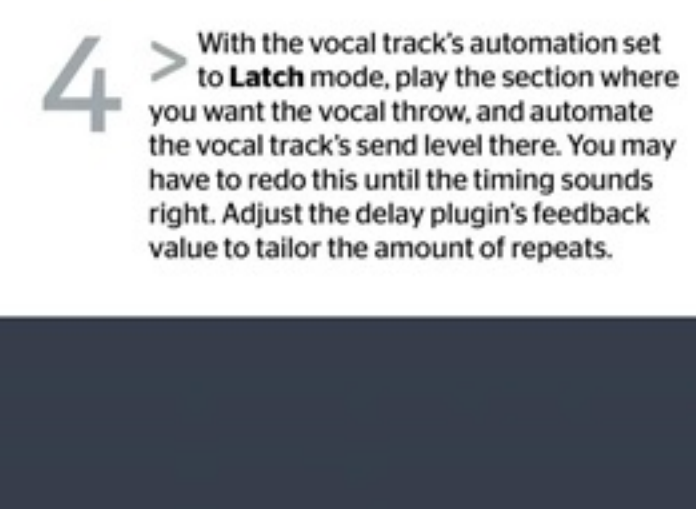
3 > Next, configure a send from your vocal track to send that track's signal to the aux return track hosting the delay plugin. Turn up the send amount to the return track, and your vocal signal will hit the delay, then echo into the mix.

4 > With the vocal track's automation set to **Latch** mode, play the section where you want the vocal throw, and automate the vocal track's send level there. You may have to redo this until the timing sounds right. Adjust the delay plugin's feedback value to tailor the amount of repeats.

# Automating EQ

Draw in EQ settings to correct section-specific problems or create filter-like effects




1 > Automated EQ changes are handy for creative purposes, like changing the tone on a particular section; or corrective ones, like fixing harshness in a vocal sound that only happens on the chorus when the singer's register is higher. Set the track's automation mode to **Touch**.

2 > Play the first song section you want to change, move the EQ band (or whichever parameter you're automating) to the new value, then release it at the end of the segment. Repeat the process for other sections of the track.

3 > To finish, flip your DAW's automation mode back to **Read**, then reach for its automation editing features and correct any spots where your timing lacked precision – you'll want to get any EQ changes perfect in the mix. **cm**

# 10 quick tips: mixing vocals

We have a huge vocal mixing feature over the page, but here's a quick fix

01 Whether you're comping vocals via the traditional audio-editing method or using recording aids such as Logic's Quick Swipe comping, you can still end up with unwanted by-products such as double breaths, which are a dead giveaway that a vocal has been comped. Editing out all breath sounds from a vocal can make it sound unnatural, so for a more complete-sounding performance, zoom in on your region boundaries and adjust them to ensure that there's only a single breath between each line.

02 When compressing vocals, the goal is to even out the volume level so that every word can be comfortably heard. So start with a generally good overall monitoring level, turn up the ratio so that you can see the compressor's gain reduction meter moving, then set the threshold so that loud phrases show a lot of movement but quiet parts barely move the meter at all. Finally, adjust the ratio so that the vocal doesn't sound overly squashed and increase the makeup gain control to suit.

03 There are numerous plugins dedicated to the task of de-essing vocals, but if you find yourself without one, you can craft your own using a simple parametric EQ and any compressor with a sidechain input. Copy the vocal track and use the EQ to isolate the sibilant frequency, normally around 5-8kHz. Then insert a compressor across the original vocal track and send the EQed version to its sidechain input. The compressor will now reduce the volume level whenever a loud 'ess' sound occurs.

04 The classic gated vocal effect can be achieved using any software noise gate with a sidechain input. First, copy the target phrase to a new audio track and place a noise gate plugin across it. On a MIDI track, load up a default sine wave synth preset with a short release, then record or program a new MIDI part, playing the required trigger pattern with this sound. Finally, route the audio from this part to the gate's sidechain input so that it triggers the gate.

Combat mismatched vocal levels by combining light compression with volume automation

05 A combination of gentle compression and volume automation can produce great results on vocals. Set up a compressor with a ratio of around 3:1, then write-enable automation and adjust the channel fader as the track plays. Fine-tune the volume curve further if necessary.

06 As well as quantising the pitch of incoming audio to a set scale, Antares' Auto-Tune can also accept target pitch information from MIDI notes, making it behave a bit like a monophonic vocoder. Click the Target Notes Via MIDI button, then either set up Auto-Tune on a MIDI channel as a MIDI-controlled effect and set your vocal as the sidechain input, or insert Auto-Tune across the vocal track and route the output of a new MIDI track to it.

07 TAL's Vocoder is a great way to get the classic vocoder sound. To get it working, strap Vocoder across a vocal track, then set up a MIDI track and send its output to the vocoder. In Logic Pro X, it's done slightly differently, by inserting the vocoder on a MIDI track as a MIDI-controlled effect, then setting the track containing the source vocal as the vocoder's sidechain input. Then just play your MIDI keyboard as the track plays to hear the effect.

A vocoder like this one from TAL lets you use your vocal signal to modulate a carrier-like synth sound

You can straighten out excess vibrato with Auto-Tune's Adjust Vibrato feature

08 Pitch-correction tools such as Logic Pro X's Flex Pitch or Antares' Auto-Tune can also be used to iron out excessive vibrato on individual notes. In Auto-Tune's graphical mode, highlight the section you want to edit and set the Adjust Vibrato control to increase or decrease vibrato to taste.

09 Once you've comped your perfect lead vocal, you can use any half-decent unused takes as backing vocals. Compress them, pan them evenly across the stereo image, then either run the spare takes beneath the lead vocal track to fatten it up, or repitch them to make harmony parts.

10 To get the popular momentary slowdown effect, put a pitchshifting plugin across the vocal and draw a pitch drop using automation. If you have Logic, you can go one better: set the Fade Out in the Region parameter box to Slow Down and draw across the target word with the Fade tool.

# GET GREAT VOCALS

## (WITHOUT BEING A GREAT VOCALIST)

Behind pretty much every great song, hit record or memorable performance lies a great vocal. But what do you do if you, like us, can't sing for toffee? Luckily, this is what computer music is all about: using technology to boost a lack of skill! And there's plenty we can do to make a bad vocal great...

> If you are anything like us, you have taken the music technology route simply because it makes music production a lot easier. (Almost to the point that it's embarrassingly easy, but *whisper* don't tell anyone else!) We first started using and abusing computers for music because we couldn't play anything, or afford the gear to try and play. But also, we assumed at the time, that it would eventually help with one other distinct skill we lacked: singing!

The human voice is probably the most distinctive, memorable, recognisable, flexible and any other word you can think of ending in 'ble' instrument ever. But in the wrong hands – usually ours – it can also be the most awful. So we've decided to take the voice by the microphones and use technology to do something about it!

We've covered vocal recording before in *Computer Music*, but this feature is a little different. We've enlisted music technology journalist – and actually pretty good vocalist – Andy Price to approach 'getting a great vocal' from another perspective: that of employing your own tones (or lack of them) and making them great. Essentially we said to him: "write a feature about turning our bad vocal into a great one."

And with that, over the next 18 pages he covers everything you need to make your – or someone else's – vocal sound better. He's essentially breaking it down into pitch-shifting, using other vocal effects and mixing vocals, but there's a lot of other vocal ground covered along the way.

One of the main processes used over the last three decades has, of course, been automatic tuning, the 'hoover' of which is Auto-Tune. We'll explain in detail how you can use that – and many other pieces of tuning software – to create better vocals, and also to produce 'that' contemporary effect. But we'll also cover how you can keep humanity in your vocal recordings, if that is your target rather than the (arguably overused) effect used in much of contemporary chart music.

There are plenty of buyer's guides too, covering the aforementioned pitch effects, vocal booths, microphones and a lot more. And the feature closes with six pages of hands-on workshops that will hopefully have you tweaking software to tune and improve your own vocals in no time.

Laterwe also round up some of the best vocal processing plugins that we have reviewed in *Computer Music*. There's everything included there from mix plugins to tuners, multi-effects to vocoders.

And if all of that isn't enough, there's a batch of vocal samples to download with this bookazine (see the contents pages for more on these) so you don't really have to do anything to get a great vocal: just load in one or two of those!

So turn the page and turn that bad vocal into a great one!

# How to make a bad vocal good

While there are many fun ways to morph, twist and chop your vocals into a huge range of sonic shapes, what about the most straightforward of aims – simply making an out-of-tune vocal not just passable, but stellar?

For many years, since the advent of recorded music, those lacking natural singing prowess would have to work extra hard to lay down their vocal takes – recording again and again until their caterwauling had been strenuously slotted into tonal alignment. The controversial appearance of Antares Auto-Tune in 1997 not only salvaged time back from end-of-their-tether producers and panicking, self-doubting vocalists, but opened up a whole new window on how the human voice can be worked with, irrespective of natural ability.

Here in 2023, there's no shame if you can't quite reach the high notes, you're struggling with intonation or just not able to flawlessly imitate the confident character of the stars. As long as you've got a clean vocal take (from an audio POV at least), software from the likes of Celemony, Synchro Arts, Soundtoys and Waves can grant us the means to mould a sparkling vocal from even the most atonal of takes.

## Wrangling the raw material

Before we delve into how software can help, it's worth stating from the outset that even the best vocalist on earth can sound utterly soulless recording with inadequate gear. While your tone, pitch and frequency can be shifted after the fact, it's critical to make sure that the clarity of the vocal recording is as clean as possible. Capturing your vocal, with a mind to working on the take further in software, needs to entail making sure the recording space is as acoustically 'dead' as possible. We want to limit reverb at this stage (as it's uncontrollable when hard-baked into the recording). Positioning your microphone at the right distance from your mouth for the general volume of the type of vocal track you're laying down is also a top priority – if you're going to be belting out a raucous rock track, you'll want to keep some space between your mouth and the grill, while closer-mic'ed vocals are more malleable for softer-toned genres. Keeping tabs on your input gain, and avoiding hitting the red is even more essential than usual when you're recording your vocal with a mind to using it as the raw material for software processing.

> "In 2023, there's no shame if you can't reach the high notes or flawlessly imitate the stars"

There's a whole lot more nuanced considerations when vocal recording of course, though that would be worthy of an entirely separate feature. We're going to assume then, that your vocal take has been captured as spotlessly as possible, in an acoustically suitable environment. For those lacking the 'X-factor', here's where the nitty gritty begins.

## Crank my pitch up

Ever since Antares Auto-Tune's launch nearly 25 years ago, we've seen a wide-range of software that not only provides the means to hard-snap to a key, but also facilitates gradual, precision

tuning. This is an essential thing to note: Auto-Tune in a technical sense is the automation of pitch correction, while pitch correction software as a whole can enable you to adjust the sharpness, or flatness, of your note to the required position. The more regimented global application Auto-Tune has ceased to be regarded as a route to perfecting duff vocals, and instead has now morphed in the minds of many into a creative tool that engenders the type of vocal experimentation that we hear in the likes of tracks by Billie Eilish, Akon, Stormzy and Radiohead.

Celemony's Melodyne 5 is widely hailed as one of the most essential pitch-correcting powerhouses. Melodyne requires you to import the sections of the stem that you wish to edit (or, as one long vocal take). Detecting the notes within the audio, Melodyne displays a series of red and orange blobs – representing the chromatic area, snapped to a grid of keys, with your vocal information rendered as a line within. This visualisation allows us to see just how badly out of tune we are.

Using Melodyne's 'Main' tool, you're able to manually heighten and lower the pitch at those specific points that need attention. Better than that, any out-of-time sections, or inconsistent note-holding can be snapped to the grid, or dragged to fit the length of time you wish it to be held for. While you may imagine that the end result will be a clunky mess, Melodyne is renowned for not showing these joins.

While Melodyne is often put on a pro's pedestal, and is now integrated within the architecture of Pro Tools (in its 'Essential' iteration), this same principle can be sampled by using Logic's Flex Pitch or Cubase's VariAudio. Flex Pitch can be used as either a global 'up-tuner', or a manual pitch correction chisel. VariAudios' approach is reminiscent of Melodyne. Both allow deep editing controls, such as the adjustment of the amount of pitch variation within the stem, manipulating the levels of vibrato present, and such factors as aligning how far your general pitch is naturally drifting away from where it should be.

With tools such as these, it's now no longer an impossibility to gradually carve your vocal into something that resembles something delivered by a vocal powerhouse.

## Keep the humanity

Ever considered that maybe your imperfect vocals don't really *need* to be that perfect? Depending what kind of track you want to make, you'll need to consider where you want to sit on the gradient between tightly-locked Cher-esque Auto-tune, and your raggedy, untreated voice. Leaving your rougher takes uncoloured could possibly add that more human element that your track might have been missing, without you even realising. If your vocals definitely need work however, then it's critical that you don't go too far down a pitch-correcting rabbit hole, and end up with a track that sounds overtly robotic (unless that's what you're specifically after). If you want to just smooth the harsh edges, yet maintain a raw edge, avoid doing anything globally. You'll need to tackle those issues more surgically. Only adjust those notes that are absolutely screaming for it a few cents in the right chromatic direction. Once you've got a version of the vocal that works, bounce it down and use that as a benchmark on which to comp further takes – this can help you get to where you need to be, without losing your vocal personality, and avoiding the sonic equivalent of the 'uncanny valley'.

Logic Flex Pitch gives an accessible way to fix pitch issues

# Vocal microphones

## Aston Microphones **Spirit** £285

The release of the Spirit marked the moment when the industry suddenly listened up and took proper notice of UK-based manufacturer Aston Microphones. With a triple-state switchable polar pattern and large diaphragm condenser, the Spirit delivers crystal clear vocals with ease. Its durable build means it's ready for regular use.

**www.astonmics.com**

## Lewitt **LCT 440 Pure** £220

A true condenser vocal superstar, Lewitt's 440 Pure has garnered giddy acclaim for its capacity to handle the human voice – from impactful rap to soaring falsetto. Its gold-sputtered 1" capsule and shock mount ensures faultless audio capture, with its gold-plated XLR connector and design showing off its build quality.

**www.lewittaudio.com**

## Shure **SM7B** £299

The undisputed gods of workhorse microphone design, Shure's SM7B presents all-round vocal dynamite. Whether you're wanting to speak, rap, sing or host – the SM7B's air suspension shock isolation and internal pop filter eliminates any unwanted interference, leaving just your vocal to sparkle as intended. The classic cardioid pattern rejects any off-axis audio.

**www.shure.com**

## Audio Technica **AT2020** £89

For those who are on a budget, the quality of vocal recordings you get with Audio Technica's AT2020 has to be heard to be believed. With its heart-shaped sound capture area, this cardioid condenser presents an isolated and accurate voice. It's exceptionally well-balanced and can also serve as an all-rounder in the studio.

**www.audio-technica.com**

## Røde **NT1** £190

Available as a bundle pack, complete with pop filter and XLR, the NT1 brings a vintage character to vocals, despite its ultra-modern internals. The mid-range response of the NT1 is particularly notable, while extremely low handling noise and shock-mounted capsule guarantees no interruptions in its job, to help give your voice clarity and focus.

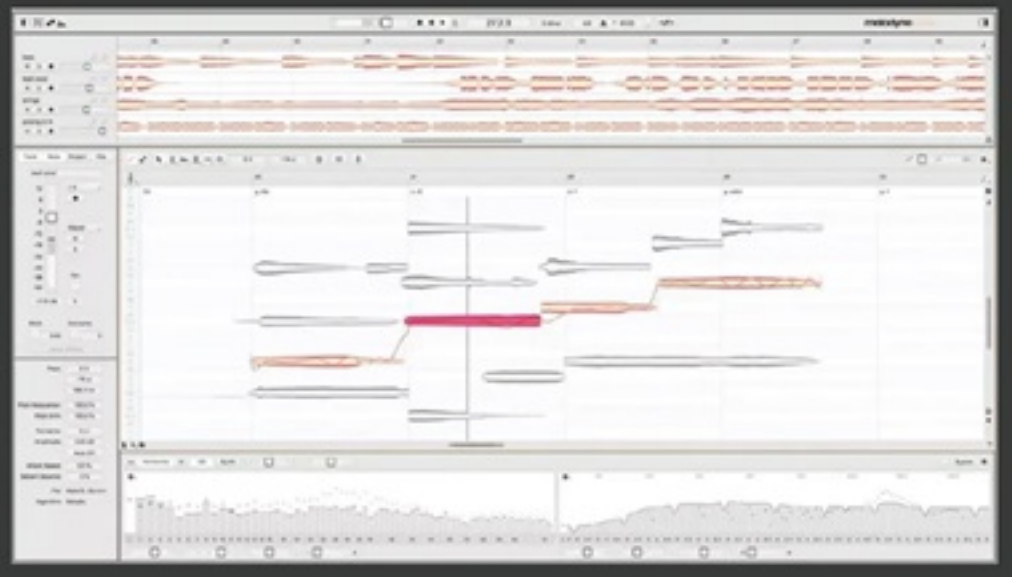
**www.rode.com**

## IK Multimedia **iRig Mic Studio** £140

IK's mobile-friendly pro studio mic is surprisingly robust for its size. Its low-noise, high definition preamp, onboard gain control and large-diaphragm capsule bely its small size, and let you grab your voice, wherever you are, with absolute class. Its diverse array of connectivity options mean it's not limited to one studio setup either.

**www.ikmultimedia.com**

# Pitch correction software



## Celemony **Melodyne 5.1**
### Mac/PC From £82

The ultimate pitch-correcting software, Melodyne's very clever DNA technology presents a uniquely detailed way into your vocal recordings, providing the means to carefully nudge the pitch and intonation of your voice, and also create doublings, harmonies and formants. It's the real pinnacle of vocal adjusting tools.

**www.celemony.com**

## Zynaptic **Pitchmap**
### Mac/PC £260

Zynaptiq's real-time wonder allows for the editing of pitch while it's being played, correcting any tuning issues on the fly. This could be a great choice for live performance assistance, or to use as a vocal-bending instrument for melodic writing. You can also use a MIDI keyboard to define a new musical playing field for your voice.

**www.zynaptiq.com**

## Logic **Flex Pitch**
### Mac only Included within Logic Pro (£174.99)

Logic's in-built pitch corrector allows you to edit the position, timing and length of notes in much the same way as a piano roll editor, while also affecting its qualities by adding vibrato or gain to the vocal take. The characteristically simple interface is easy to grasp if you're au fait with Logic.

**www.apple.com**

## Waves **Tune**
### Mac/PC £30

Waves' modern classic has a huge user base, with some even preferring it to Melodyne. It has a similarly broad assortment of vocal tweaking tools, with its waveform overview, piano roll and pitch editor, as well as three timeline modes. The Waves Tune family also includes Real-Time, which can instantly re-work live vocals.

**www.waves.com**

## iZotope **Nectar 3**
### Mac/PC £107

The masters of audio surgery are also a dab hand when it comes to vocal bolstering. Nectar 3 isn't just a pitch-correcting powerhouse, but assists with vocal mixing, polishing and production. With vocal accentuating unmasking tools and 12 vocal processing modules, Nectar 3 is a fluid route to achieving vocal excellence.

**www.izotope.com**

## Antares **Auto-Tune Pro** Mac/PC £530

The original, and some say, the best – Auto Tune Pro has evolved far beyond the initial bad rep for heavy-handedly applying rigid vocal tuning, and is now a nuanced, manual-editing beast. While more than capable of being used to form Cher-like pitch leaps, this latest version is jam-packed with humanising features and tools.

**www.antarestech.com**

# Processing your vocal

While fixing of out-of-tune or messy vocal performances is an achievable aim, tuning is not always the central issue with vocals. There's an array of routes to make your voice a far more interesting sonic element of your mix

Pitch-correcting and Auto-Tune software can help to massage and sculpt a duff performance into something that shines. But that's not necessarily something that every track calls for. It's become a truism that vocal manipulation – be it formant shifting, stuttering/glitching, extreme compression, distortion or rigidly aggressive use of Auto-Tune to replicate a sort of vocoding effect – is so ubiquitous in today's pop landscape that it's more unusual to find a track that *doesn't* exhibit it in some degree.

Creative vocal effects can lift certain moments of a track, bolstering not just the impact of a vocal performance. Effects and treatment can add dirt, space-age surreality, transform the character of the singer entirely, scaling between a sinister-voiced demon or a high-registered chipmunk, or re-imagine the vocal as bendable instrument, leaping into ranges that no natural human voice could.

## Ghost note rider

Aside from these extreme swings, it's common practice, irrespective of genre, to rely on subtler automation of volume, gain, and for spatial effects like delay and reverb. Harnessing automation to scale with the dynamics of the mix allows for your vocal characteristics to sonically adjust to peaks and troughs.

While now managed by your DAW, this principle stems back to the hands-on days of fader riding, whereby a recording engineer would manually scale the gain of a vocal recording live. As the singer delivered a take, the engineer would adjust the gain level of the input signal, responding to the volume of the vocal. You can still control your gain this way today, in software, by using the likes of Waves Vocal Rider or TB Pro Audio GainRider 3. In terms of wider vocal mix automation, such as applying filter sweeps, delay, reverb levels and more, most of today's DAWs offer the means to latch your

**Waves Vocal Rider mimics a manual studio action**

movements across multiple plugins as the track plays back, recording your setting tweaks and repeating those steps as you go. You can also create auxiliary channels containing the required effects, sending the volume from them at certain predetermined points.

**Waves Ovox offers a slew of creative effects**

## Mouth monster

Now we've got a handle on how we can manage our vocal's broader characteristics from a mix management point of view, and before we take a walk on the wild side and explore some particularly gnarly vocal effects, let's take a look at some calculated use of EQ filters. Using high-pass filters in dramatic ways can impart a radio-type, low-frequency quality to your vocal which you probably don't need for the whole take, but can colour certain areas of your mix. While your basic vocal EQ should be attenuated to eliminate anything below 100Hz, by increasing the attenuation to nothing but the highest frequencies of your vocal, you can conjure a thin-sounding, razor sharp vocal effect that periodically emerges at key points of your track. The reverse is true of automating a murky low pass filter at a certain point.

While extreme mixing is one thing, there's also a slew of products that deliver a huge range of colourful vocal effects instantly. Waves OVox is one particular plugin effect with the power to contort your vocal dramatically. "OVox is literally playing a synth into your vocal envelopes, and both the synth and the envelopes can be further tweaked and modified." Waves' Or Weisinger explains. "You can add special effects and orchestrate the entire plugin using the different modulators. You can also replace the built-in synth carrier by any other audio and experiment with uncharted territories. The results are usually everything other than a polished vocal!" Check out our buyer's guide for our choice of vocal-mangling tools.

## Stabs and stutters

Beyond the shaking up of that lead vocal, what about using additional vocal recordings as a flexible, tone-shifting supplemental instrument? These kinds of sounds are often the foundation of trap, house and techno arrangements. It's easy to find vocal stab samples around the web (and in fact, you can find quite the free collection over at musicradar.com/news/tech/free-female-vocal-samples). Using these raw samples as a starting point, you can use your DAW's sample editor, such as Logic's Quick Sampler, Cubase's Sample Editor or just Ableton Live's main sample/resample-oriented workflow to map the vocal onto your controller, allowing you to glide between pitch-adjusted variants, playing and performing the vocal stabs as if they were a synth patch. You can adjust the legato (smoothness of note transition) between keys and which note plays which pitch. Once you're ready to go, you can begin dabbling in using your vocal samples to create harsh stabs, morphing and winding vocal backings, chords and – if your Glide is set to a harsh cut-off – sharper rhythmic stabs and stutters.

# Sing sang sung

While Cher's *Believe* was notably the first example of extreme digital Auto-Tune used in a chart-smash, vocal manipulation in the context of pop goes back a long way. Even prior to Kraftwerk's pre-eminent use of the vocoder, way back in 1939, Bell Labs' Homer Dudley created the 'Voice Operating Demonstrator' – essentially a synth for the human voice that could limit the bandwidth required for military communications in World War II. Though he wasn't aware at the time, he'd created the prototype of the vocoder. These principles were applied by such pioneers as Robert Moog and Wendy Carlos on their influential *A Clockwork Orange* soundtrack, and by the BBC's Radiophonic Workshop in the '70s. But, even back in the '60s, groundbreakers like The Beatles pushed the vocal envelope in pop. On the fab four's epic closer to 1966's *Revolver*, John Lennon's vocals were run through the Leslie speaker of a Hammond organ, creating a shimmering tremolo effect. Just one element that made *Tomorrow Never Knows* so innovative. Slap delays, wrought from tape bouncing, were also common in this period, and added depth to a vocal's timbre.

# Vocal multi-effect plugins

## iZotope **Vocal Synth 2**

### Mac/PC £150

With five bendable, infinitely manipulable modules, Vocal Synth 2 presents one of the most focused ways to re-shape a vocal into something colossal. Its advanced spectral display responds to the characteristics of vowels, and reacts to your effect choices. It's a sure-fire route to instant vocal splendour, or ragged edge.

**www.izotope.com**

## Soundtoys **Little AlterBoy**

### Mac/PC £75

Garnering huge respect for their exemplary plugin range, Soundtoys are dab hands when it comes to vocal adjustment. Specialising in formant shifting, pitch adjustment and melodic tweaking, Little AlterBoy is a one-box pathway to a range of cool new vocal worlds, you can even use it as a hard-tuner, and layer your track with authentic saturation.

**www.soundtoys.com**

## Klanghelm **SDRR**

### Mac/PC £20

If you want your vocal to ripple with the burning fire of classic tube saturation, you're going to want to invest in something like Klanghelm's SDRR. Harnessing its four different modes based on vintage outboard gear, you can create complex, rich saturation and deep stereo width enhancements, leading to a hardy vocal character.

**www.klanghelm.com**

## Antares **Mutator**

### Mac/PC £55

Leaning more into the 'I want to sound like an alien' approach, Antares' Mutator is a sound design tool that can carve your voice or vocal into a modulated, pitch-mutated extraterrestrial. Deep automation effects allow you to get more nuanced while real time processing gives direct feedback. It's a versatile addition to any vocal tweaking arsenal.

**www.antarestech.com**

## Acon Digital **Multiply Chorus**

### Mac/PC £FREE

You can't really go wrong with free swag, and for no outlay, Acon's Digital Multiply Chorus can pay dividends for your vocal. Processing your voice with a phase randomising filter, it can simulate the effect of multiple vocalists playing the same tone simultaneously. There's a heap of scalability controls, too.

**www.acondigital.com**

## Waves **Ovox**

### Mac/PC £40

A vocal-twisting powerhouse, Ovox goes beyond a conventional vocoder to result in a multihued software voice-shifting synth like no other. With its classic talkbox, and artist-tailored presets, Ovox's dual high-resolution engines grant a huge scope to totally transform your voice, all rooted in the built-in 8-voice polyphonic synthesiser.

**www.wavesovox.com**

# Vocal mixing plugins

## Oeksound **Soothe 2**
### Mac/PC £119

Oeksound's smart dynamic resonance suppressor is one of the finest plugins of the last decade. Fact. Using astoundingly programmed clever algorithms, Soothe 2 can isolate problematic frequencies and warring resonances on the fly, and trim them out before you even notice they're there. It can also take away the harshness of a close-miked vocal with ease.

**www.oeksound.com**

## Valhalla DSP **Vintage Verb**
### Mac/PC £40

To work with Vintage Verb's epic room, ambience and concert hall reverb can make you feel like you're delivering the hefty top-line to a massive power ballad. Taking its cues from the hardware digital reverbs of the 1970s and 80s, Vintage Verb's incredible range of spatial tones can lift even the flattest of vocal takes up directly into the stratosphere.

**www.valhalladsp.com**

## Universal Audio **1176 Classic Limiter Collection**
### Mac/PC (£175)

Now a decade old, Universal Audio's 1176 emulation was among the first examples of the startling faithfulness of UAD software's design. Recreating a true studio legend, the 1176 collection lets you track and mix with the same gain, distortion and compression as that one-time staple of hardware.

**www.uaudio.com**

## Sonnox **Toolbox VoxDoubler**
### Mac/PC £30

VoxDoubler is focused on one of the most recurrent vocal mixing issues: enhancing a weedy vocal take. Doubling the vocal to make it sound either more thickened or spread out wider in the stereo image is the dual-pronged route that Voxdoubler takes. It's an extremely slick way to get impactful vocals quickly.

**www.sonnox.com**

## FabFilter **Pro Q 3**
### Mac/PC £130

One of the most beloved mixing tools around today. FabFilter's Pro Q 3 is the last word in pro equalisation, and is an effective sculptor of tonally sublime vocals. Its GUI isn't just pretty, its deep spectrum analysis allows you to easily apply notching, transient taming, and dynamic band EQing to your mix.

**www.fabfilter.com**

## Soundtoys **Decapitator** Mac/PC £159

Another entry for Soundtoys, Decapitator can apply some sumptuous and authentic hardware-modelled saturation to your vocal. Once the secret sauce behind truly legendary vocal takes, Decapitator makes it easy to mould a classic-sounding top-line, or can be cranked up to unleash some fiery overdriven textures.

**www.soundtoys.com**

# Mixing your vocals

Though you can correct the pitch, tuning and vibrato of your voice in software, as well as contort it into all manner of extraordinary sonic shapes and characters, it's all for naught if you can't actually *hear* your vocal in the mix

Regardless of how tightly in-tune your lead vocals, or how bizarre and idiosyncratic the treatment you apply to it, if a colossal synth lead is daubed all over it, or if pounding sub-bass renders it a mere distant echo in your mix, all that hard work will ultimately still lead to a disappointing, ineffective vocal. Therefore, it's critical we take the time to listen to our vocal within the overall sound of our mix.

A primary question we need to ask ourselves from the get-go is whether we want our vocal to be the top-line element – as is traditional in most contemporary genres. We're going to assume that this is the case, though we'll also look at effective mixing of vocals-as-instrument later in this 'mixing' section of our vocals feature.

## Roll deep

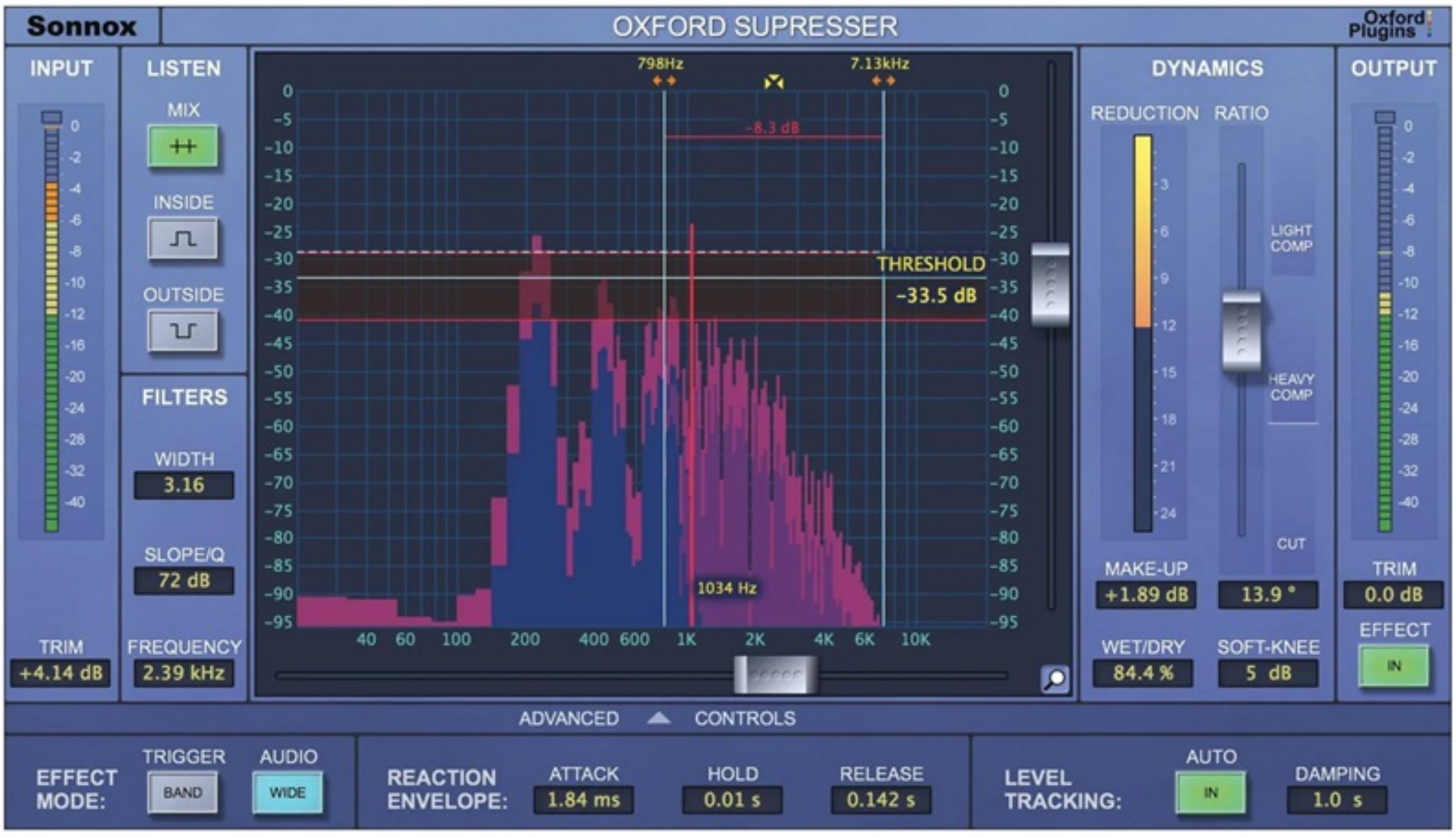
By the mixing stage, you should have already comped the best performances into your final mix element, and used the appropriate amount of pitch-correction. Whether you've run the vocal through a vocoding or effects plugin chain isn't massively pertinent at this time. What is fundamental to take into account, is whether your vocal is rubbing up alongside other instruments within the mix.

> "It's critical that we take the time to listen to our vocal within the overall sound of our mix"

As we explained in the previous section, using a high-pass filter on frequencies sitting between the 80Hz to 150Hz range will eliminate the rumble of extreme low end, as well as that all-too-common vibration that comes from things such as passing traffic when recording. Though we emphasised that some aggressive high pass filtering can concoct odd vocal characteristics, in a general mix context you should apply caution at every turn. Low-end roll off is fairly easy to apply even before you get to your DAW – most microphones have a low-end roll-off switch that physically eliminates those intrusive low end sounds. Though this can be instantly helpful, be mindful of how your vocal sounds with this more extreme barrier applied. You don't want your vocal to sound too unnatural, so more selective low-end notching in software may be preferable.

## Breathing room

With your vocal now a punchy mid-frequency range aspect of your mix, you're going to need to be very mindful of your stereo image, and the instruments that compete with your vocal for attention in the same frequency space. Guitar, piano, violins and synths are particular culprits here. While the high-pass filter serves to build a great barrier which keeps the low frequencies out, more surgical subtractive EQing is required when dealing with extremes in the mid-high frequencies. In fact, it's often an idea to apply a frequency sweep to your vocal. Using a parametric EQ such as FabFilter Pro Q 3 or iZotope's Neutron, we can sweep across the frequencies and spotlight serious problems. We set the band frequency to somewhere around 30Hz, with around 10dB of gain. Once you play back your track, gradually raise the peaking filter frequency across all the frequencies of the track. When you notice the particular problems




**Sonnox Oxford SuperEsser stops your take sounding hiss-terical**

**Waves Sibilance is another hero for saving us from the agony of unwanted vocal noises**

getting better (or worse) then stop and zoom in – you've found the area which you need to either reduce or boost.

Compression is another major factor to consider when mixing vocals. Though typically used as a means of reducing or enhancing gain, apt use of a compressor on a vocal can also impart space and depth to the sound. Though there are all manner of ways to approach your vocal's compression level, consistency is the general rule of thumb – you don't want extreme dips or peaks in gain levels, and they need to find a place somehow above the mix.

Taming transients is a little different with vocals, as – unlike drums – there's no 'hit' to tackle; instead, we're mainly dealing with plosives, consonants and other lip-smacking sounds. More delicate use of EQing, compression and de-essing plugins such as Waves Sibilance and Sonnox Oxford SuperEsser can eradicate those unexpected vocal tics and lingering 's' sibilant sounds. Careful use of all of the above is key to moulding a vocal that has both clarity and punch.

## Chopscotch

Aside from the lead vocal, backing vocals or vocal stabs/chops need to be approached with a very different, more mix-considered mindset. Chops in particular, are essentially sliced-up samples of vowels, typically played back in new rhythmic shapes. Therefore, they generally need to be considered not so much as standout mix components that battle with your lead vocal, but supplement it by being placed at less conspicuous corners of the stereo field. Often, vocal chops can be emphasised by either extreme pitch-correction and warping (to make them sound robotic) or daubed in reverb (to give them more natural sounding space).

*We've now explored how pitch-correction, effect processing and mixing can enhance even the most amateur of vocalists' takes. Ultimately, with the gamut of software solutions available, lacking natural ability is no reason to feel a lack of confidence about working with your voice. Now we'll explore some vocal hardware and get hands-on with some software workshops.*

# Choirmaster

While lead vocals and supplemental vocal chops and stabs are to be treated as individual mix components when marking out your frequency space, when it comes to larger-scale vocals within a track, such as when you're juggling several backing vocals to craft something akin to a choir, then you're going to need to be extra careful that your voices coalesce, and don't sound disjointed, or have wayward tracks spinning off into busy areas of your mix. Typically, backing vocals will always need to be quieter than your lead vocal take, particularly if they're seeking to bolster or enhance the mood of your song. With this lower volume comes a more broad brush approach to spatial effects such as reverb. Hall or chamber reverbs on multiple background vocals present a quick means to mesh them together within the same (virtual) space. It helps to use a different type of reverb, and different vocal treatment altogether for that matter, than the lead vocal. Placing your backing vocal tracks in one specific area of the mix (as opposed to panning them around) aids the perception that you've actually recorded a small group of human beings for real. If reality is your aim then, don't over-do the pitch-correction; it sounds more organic.

# Best interfaces for vocal mics

## Universal Audio **Apollo Twin Mk II**

**Mac/PC From £850**

One of the world's most acclaimed interfaces, the Apollo Twin MK II is a real winner for vocal recording, as well as anything else you might throw at it. With internal DSP handling the processing strain, it's easy to enhance even a dry vocal take with UA's Unison modelling technology.

**www.uaudio.com**

## Focusrite **Scarlett 4i4 3rd Gen**

**Mac/PC £175**

Focusrite are dab hands when it comes to affordable studio quality interfaces, and the Scarlett series is among the industry's most ubiquitous I/Os. Now sporting an Air switch – giving more high frequency space and clarity to the vocals that it captures – the 4i4 is an affordable route towards awesome vocal takes.

**https://focusrite.com**

## Steinberg **UR44C**

**£240**

Steinberg's I/O powerhouse is more than ready to handle the rigours of a full-band recording. But it is particularly adept at vocal capture, thanks to its D-PRE microphone preamps, delivering warm, detailed and transparent reproduction of the human voice, with zero latency. Even if you're not a Cubase user, the UR44C is certainly worth a test.

**www.steinberg.net**

## Native Instruments **Komplete Audio 6**

**£209**

Software titans NIs' foray into the world of studio interfacing is a triumph. The best of these is Komplete Audio 6, and features six pristine in and six-out connections, ultra-low latency, 48v phantom power and easy to understand LED VU meters to keep tabs on signal strength. It's a real beauty.

**www.native-instruments.com**

## Audient **iD4 Mk II**

**£105**

Audient's iD4 makes a serious case for being the best interface that money can buy, especially in the vocal domain. With an Audient console hybrid discreet mic pre and class-leading converters within, every vocal take is captured with beautiful accuracy, and with low noise and classic analogue warmth. Top drawer stuff.

**www.audient.com**

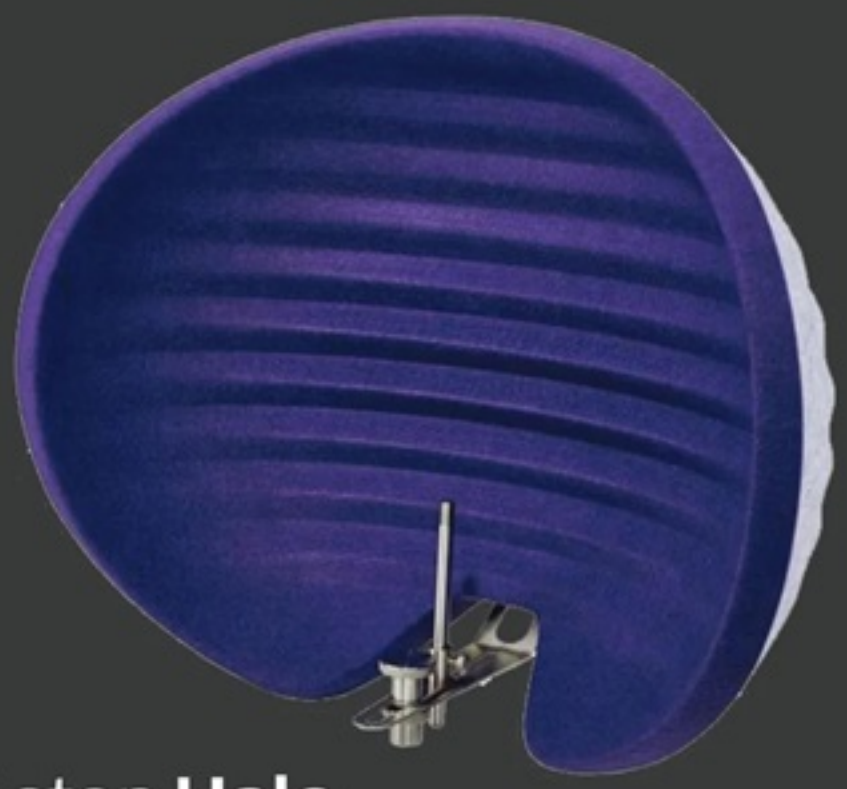
## Apogee **Duet 3**

**£499**

This modern version of Apogee's renowned Duet interface sports hardware DSP underneath its tiny hood, as well as two world-class preamps. The DSP is controlled by a software channel strip, tuned by American classic rock studio legend Bob Clearmountain himself. It's the ultimate vocal hub.

**www.apogeedigital.com**

# Vocal booths and accessories



## Aston **Halo**
**£205**

A portable vocal isolation booth without peer, Aston's Halo re-imagined the concept from the ground up, creating a supremely effective tool for isolation, absorption, diffusion and 360 degree filtering. Its PET felt design and ease of mounting make it a no-brainer if you want completely isolated vocal capture. It looks pretty cool, too.
**www.astonmics.com**

## SE **Reflexion Filter Pro**
**£158**

SE really established the benchmark when it comes to portable vocal isolation, and their Reflexion Filter Pro is still very much a top-tier vocal isolating solution to this day. With its multi-layer acoustic fabric design, it works to eradicate intrusive room noise and keeps everything squarely focused on your vocal.
**www.seelectronics.com**

## GIK Acoustics **VISO Booth** £175)

Harnessing their many years of experience as acoustic treatment specialists, GIK's VISO Booth is both an aesthetically pleasing and technically brilliant tool for isolating the voice. The company uses exactly the same high quality rock wool core that they use in their patented room treatments when crafting the VISO Booth.
**www.gikacoustics.com**

## Soyuz **The Launcher** £150

Able to impart the character of a range of classic tones onto your mics, Soyuz's The Launcher is a nifty way to get superior quality vocal takes even if you've got a low-budget mic. It significantly boosts gain, reduces background noise and applies a secret formula to the signal path. Soyuz's device can really give you a lift.
**www.soyuzmicrophonescom**

## Kaotica **Eyeball** £189

Described as a 'microphone isolation ball', the Kaotica Eyeball is placed over your mic to keep the sounds it captures immune from room reflections, pop and intrusive external sound. Aimed at those working in untreated environments, the Eyeball can enhance and underline your vocal, keeping its characteristics in focus.
**www.kaoticaeyeball.com**

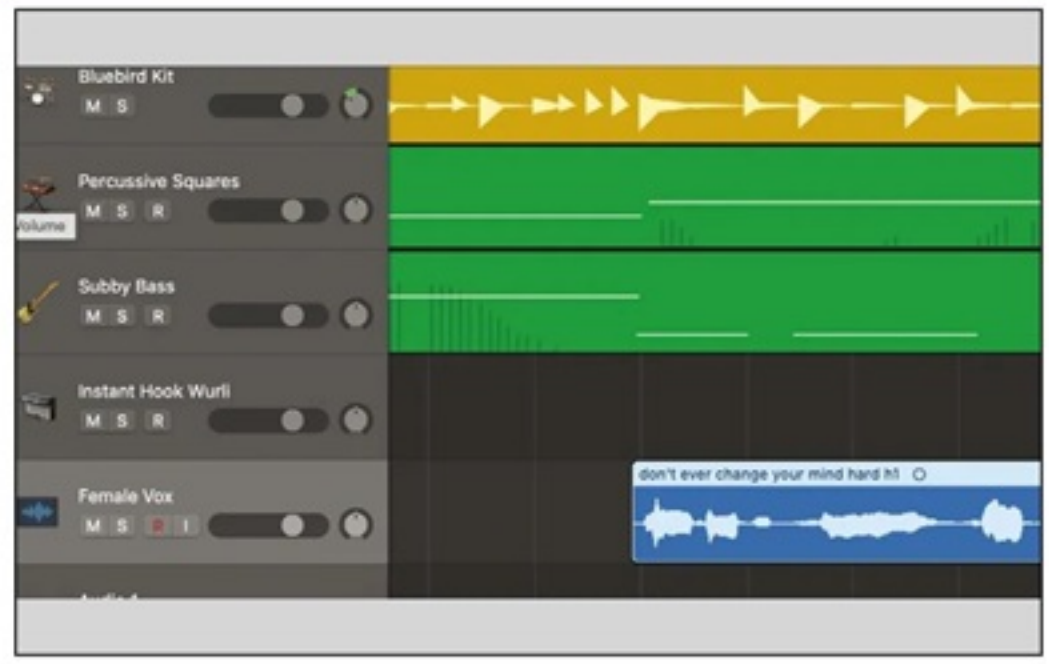
## Cloud Microphones **Cloudlifter CL-1** £145

Bringing up to +25dB extra of clean gain, the Cloudlifter's superb attributes can push your vocal performance to new heights. Harnessing JFET circuitry, the Cloudlifter is an able tool to assist your vocal's prominence in a mix in both studio and on stage. Its plug-and-play simplicity makes it all the more appealing.
**www.cloudmicrophones.com**

## > Step by step

## 1. Enhancing vocals with Antares Auto-Tune

1 > Antares Auto-Tune is a remarkably dexterous suite of voice-manipulating software, and is now capable of far more, particularly via the family of sub-plugins such as Auto-Key and EFX.




2 > We've got a demo we're working on, with a vocal melody concocted of samples. However the samples are dry and entirely out of tune and roughly in tempo with our track pitch.

3 > Opening Antares Auto-Tune, the first thing we do is snap our vocal track to the correct pitch key using Auto-Key. This addition to the Antares family intelligently discerns our root key. We can then send the key to Auto-Tune.

4 > Setting the Re-Tune speed higher can make our vocal snap more sharply to the key of the track. Cranking it right up, and reducing Humanize controls gets us something more aggressively tuned.

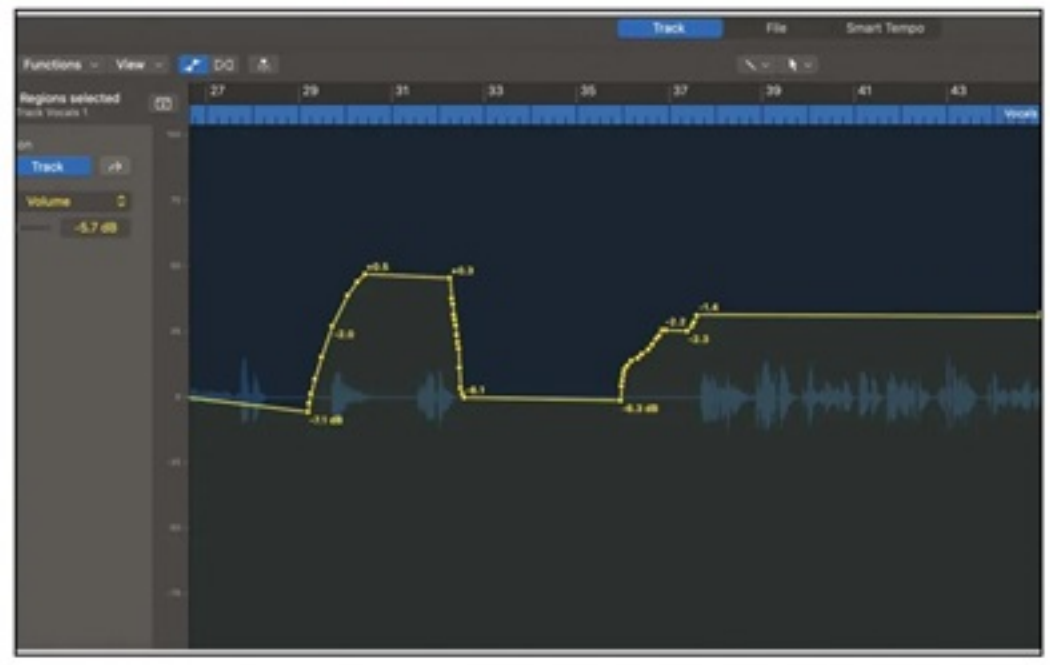
5 > Using the Flex Tune control can dial back some of the harshness for the first part of the vocal, but we want to keep a more electronic time for the second part.

6 > For this second section, we've opened the Auto-Tune EFX plugin, and applied a Duet effect. We can add up to four effects here, including spatial effects, to expand our now quite pretty vocal into something that's starting to sound much cooler.

> Step by step

## 2. Vocal mixing tips

1 > Getting our vocal to sit well in a mix often requires some delicate wrangling. Firstly let's take a look at compression. Our vocal take is bolstered by a 5:1 compression ratio, but the lower the ratio, the more we hear the natural vocal dynamics.




2 > For certain sections, we want to bolster our vocal in the mix, while other parts we need it quieter. This is where gain automation comes in. Every DAW will enable you to do this. In Logic we can raise our gain at these key points

3 > EQing our vocals requires us to listen to our vocal in context – how well is it playing with other instruments in the mix? Cut and boost frequencies accordingly. You'll want to scoop out some low end to eradicate rumble, but be careful you don't lose character!

4 > Trim out unnecessary gaps in your vocal stem, and use a De-Esser to get shot of plosives and lip-smacking. Loop a small section and isolate what frequency area is the most troublesome before applying to your whole mix.

5 > Unless you're being super surgical for a specific reason, never mix vocals in isolation. You should always find the correct gain balance and compression settings by playing all your mix elements together, avoid the solo button and hear how your track is working as a whole.

6 > Adding new effects can be dangerous if you've already painstakingly sculpted your mix. So make sure you add any extra treatment as an auxiliary channel so you can dial them in as needed and not disrupt your vocal equilibrium!

## > Step by step

## 3. Getting creative with Waves OVox

1 > Waves OVox is one of the most comprehensive vocal FX suites that money can buy, able to vocode, double, synthesise and re-wire any vocal into a tantalising new musical form.

2 > At the top left of the UI, you can select which signal to be used as OVox's modulator. We're going to choose Track, but you can also use a Sidechain input.

3 > On the OVox 1 and 2 controls, we can adjust Tune, Formant and Gain rotaries to taste, while the bottom of the UI features the mixer section. Here we can play with Voice Correction and Sibilance.

4 > Clicking the top right expansion icon reveals the hidden depths of OVox's twin oscillators. Here we can adjust deeper settings such as Note Harmonics, Noise and Formant Speed for each oscillator.

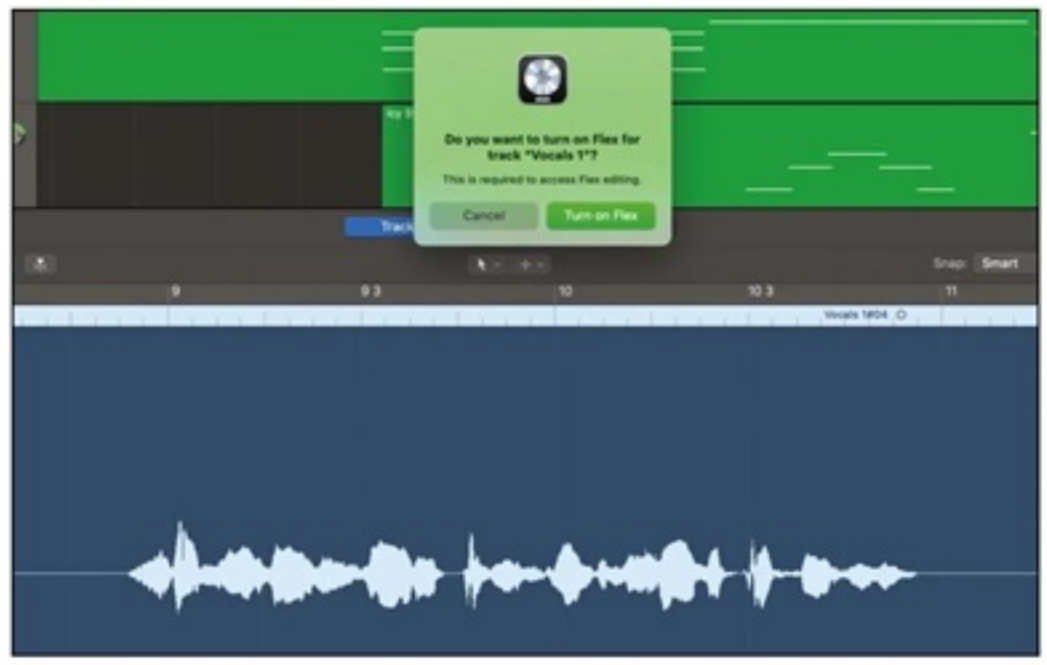
5 > It's simple enough to create a new modulator too. Dragging an LFO or a SEQ to a rotary will allow you to control the levels of the modulator. There are also up to five effect slots to further distort and enhance your vocal.

6 > Once you've crafted a new sonic flavour for your vocal sound, the central Note Mapper can force your vocal's pitch to notes, and the harmoniser can expand your single notes into chords – if you want to get ultra-robotic.

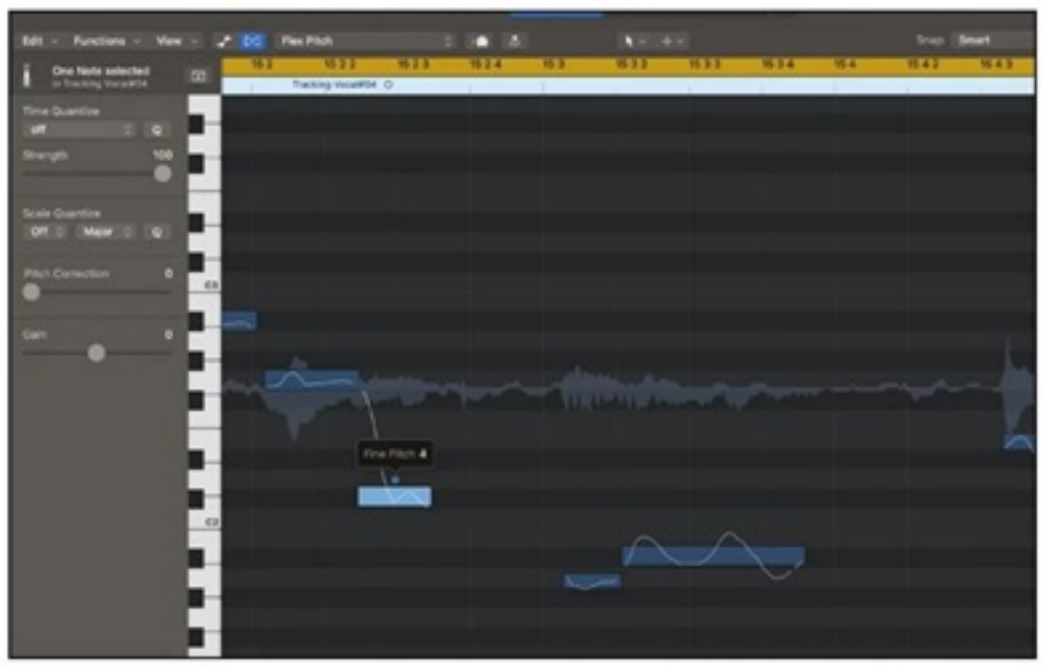
## > Step by step

## 4. Pitch correction with Logic Flex Pitch

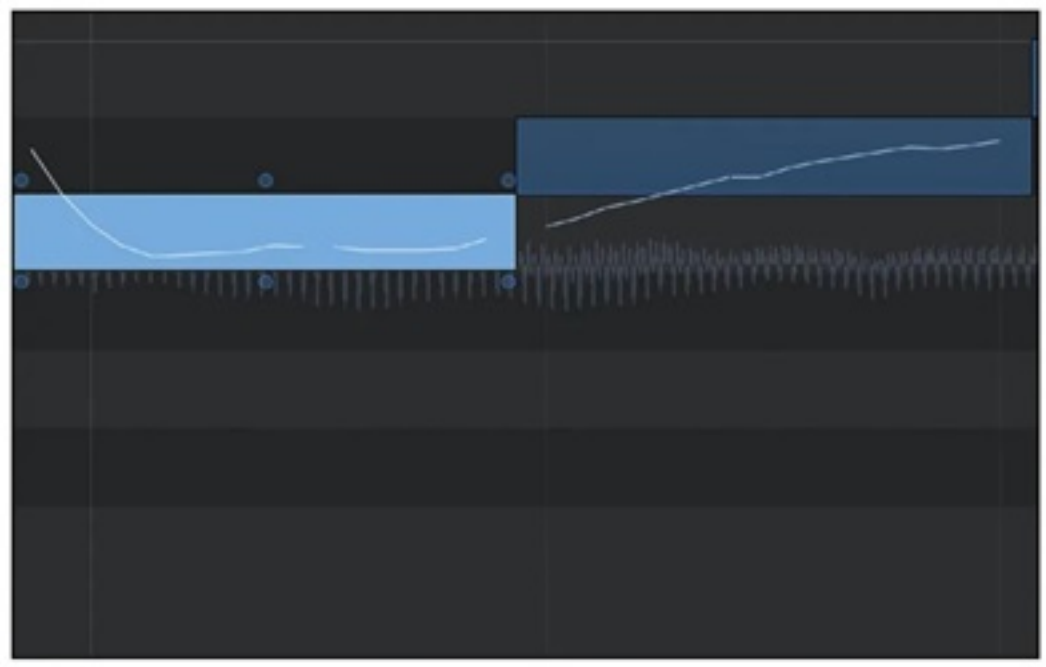
1 > We've got an idea for how we want our vocals to fit in our Logic track, matching our lead line, yet our vocal take seems a little off-pitch. Let's sort that out.



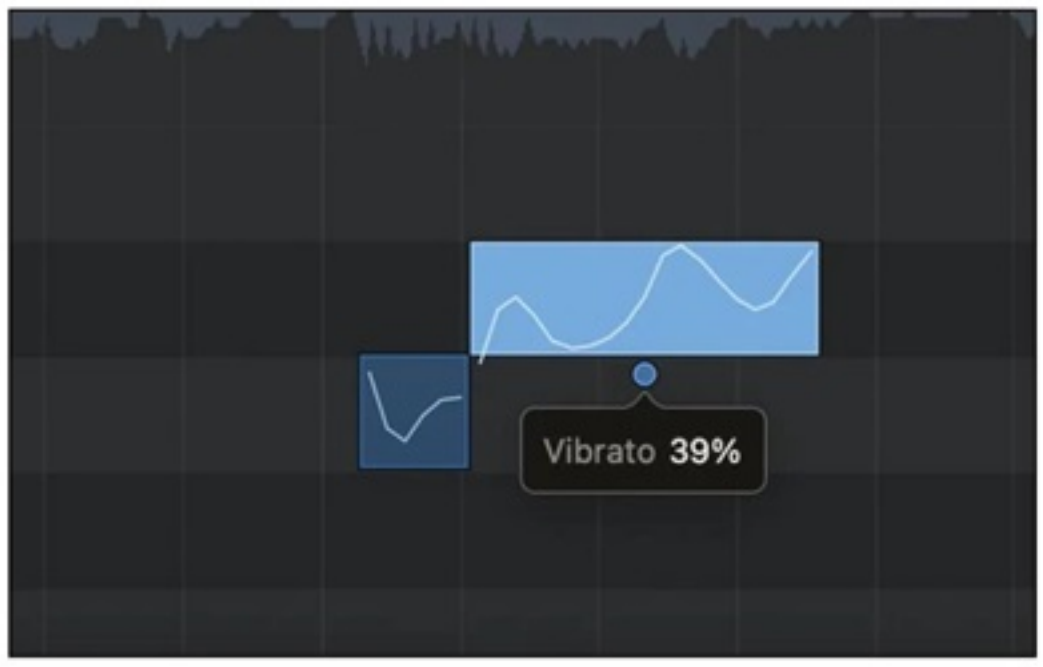

2 > Next we press E to enter the Editor for our vocal. In Editor we click Show/Hide Flex which opens our Flex options. Clicking Flex Pitch gives a visual representation of our vocal – clicking on the circles around each box indicates how far from the required pitch we are, numerically.



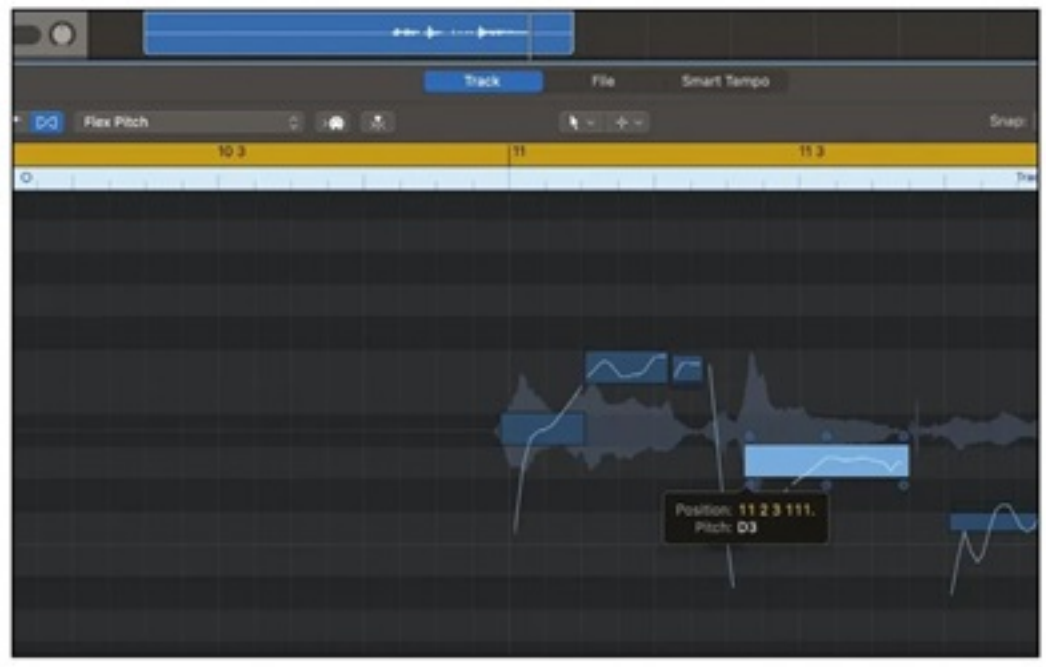

3 > It's apparent that aside from a few of our early notes, we're quite far off the required pitch – clicking and dragging the centre circle – and we now have to adjust our notes to reach the right position.



4 > We've done a broad correction to our vocal pitch – adjusting our notes to between 0 and 10 each way. However, as we can visually see, our vocal's sound really isn't a consistent pitch. The waveform drifts way out of the required boxed-in pitch area.



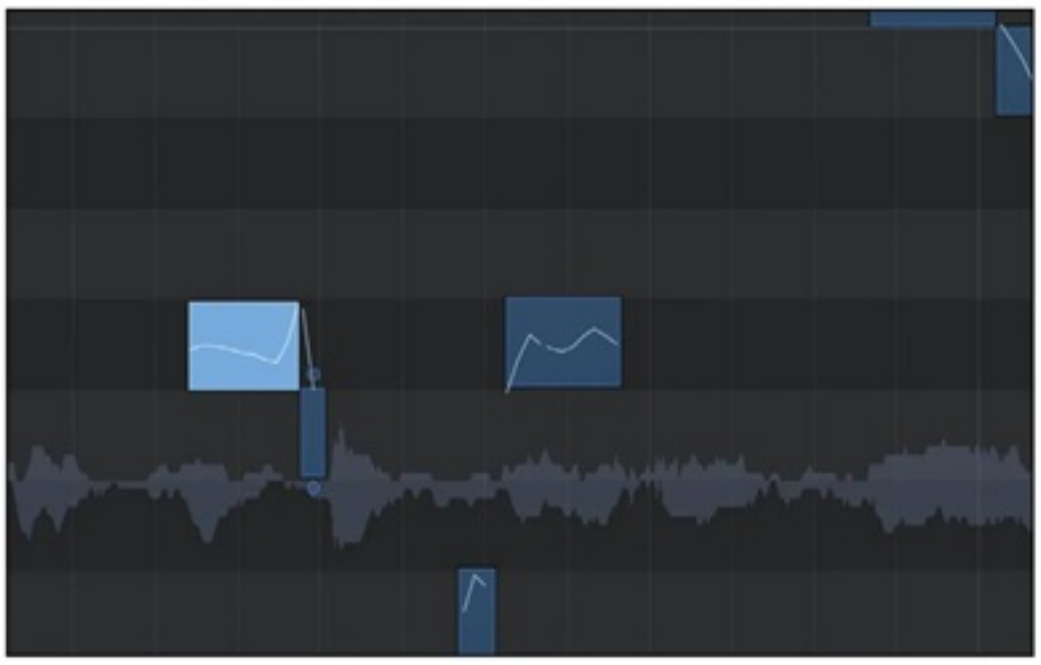

5 > Clicking the top right and left allows us to reign in this 'pitch drift' – reducing the natural off-pitch vibrato and swooping. Clicking the bottom centre Vibrato circle allows us to further mould the pitch and reduce wobble.



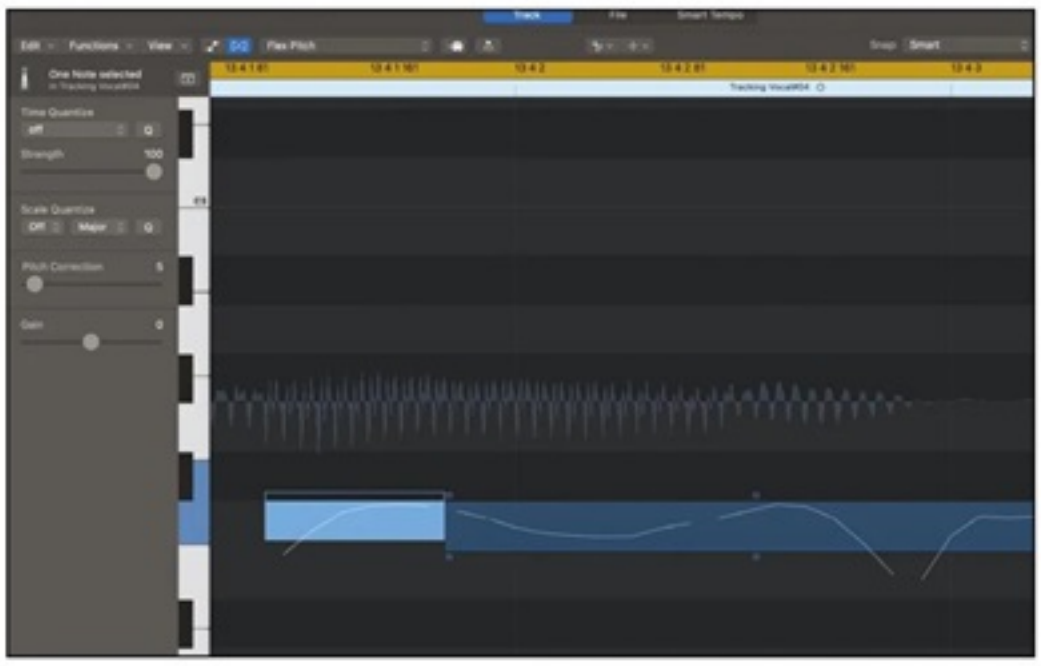

6 > It's also an idea to carefully trim down any elongated notes by grabbing the side of each box and incrementally shortening their duration. This can help eliminate any prolonged off key warbling.

## > Step by step 4. Pitch correction with Logic Flex Pitch (continued)

7 > Now our vocal is looking and sounding a little more palatable. However some parts of our vocal sound a little too 'corrected'. Keep a considered grasp on the overall sound of the voice, and don't let it get *too* robotic.



8 > There's still some problematic pitch drift. Click on the Scissors icon to make cuts to those parts of the vocal where there are more extreme issues with curves wandering out of the pitch marker area.



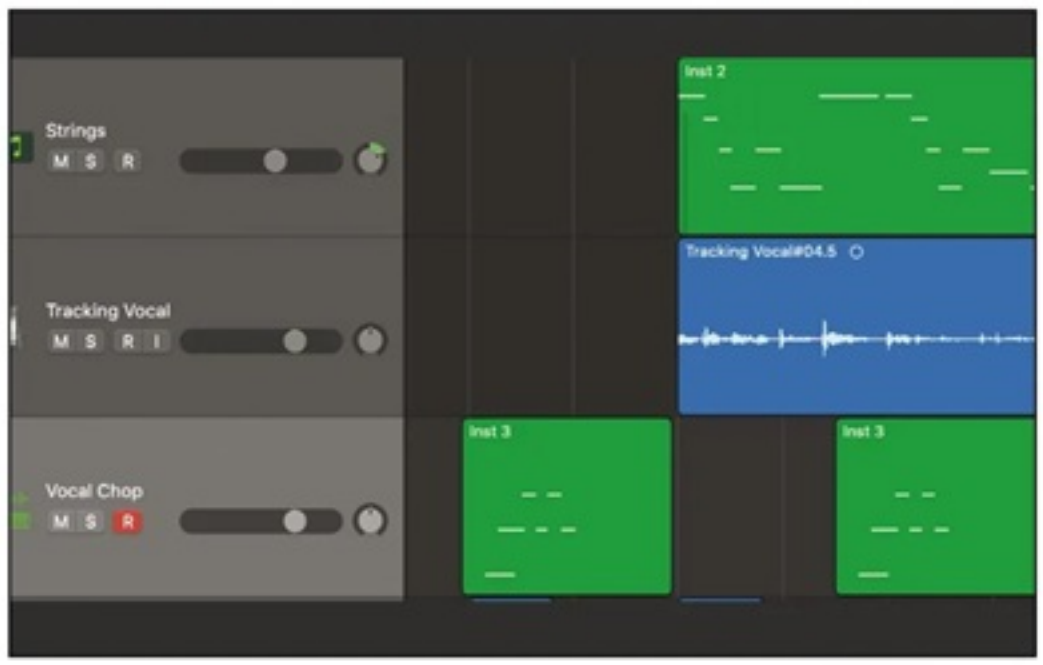

9 > Using the Scissors tool to slice out those sections creates new pitch boxes more aligned to their specific location. There's no limit to how deep into the audio you can go – but again, retain a close ear on the overall artificiality.

10 > Once you've got a finer focus on these wayward areas in your project, zoom in and use the Pitch Drift/Vibrato circles in Logic to adjust the pitch frequency to fit.

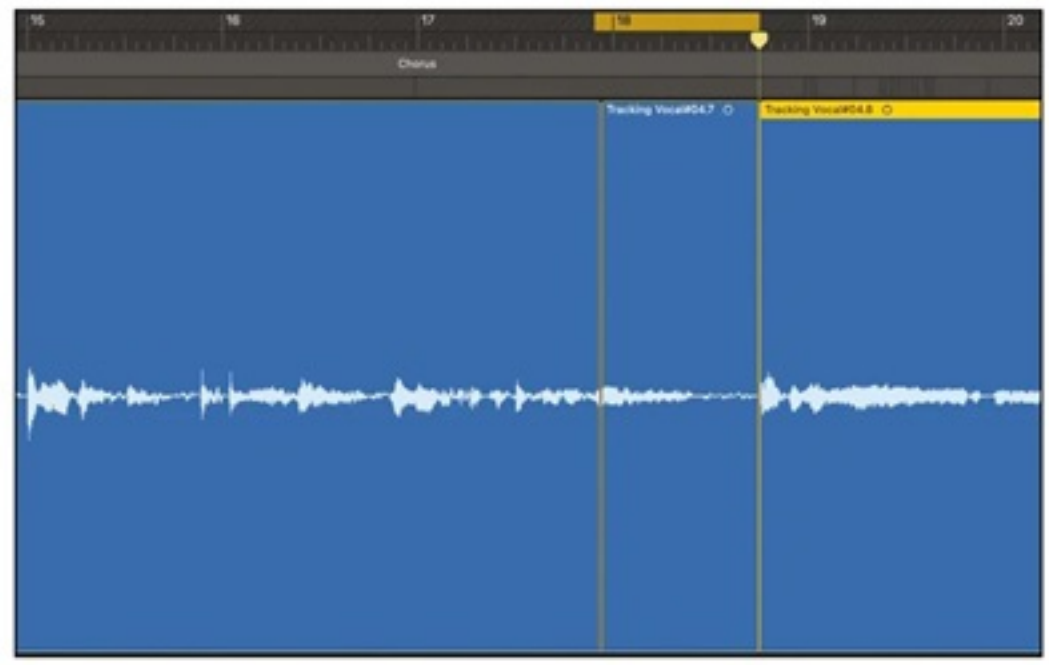
11 > Other qualities, like Gain on specific sections, and Formant transposition (represented in circles) can underline parts of the performance that weren't as prominent before, and enhance sections.

12 > While we've got some way to go before this track is finished, some adjustment with Flex Pitch has aligned our vocal to a more consistent pitch zone. How much you need to apply is down to you.

## > Step by step 5. Making a vocal sample instrument




1 > Making your own playable vocal chops, either from your own voice or from an existing sample, is a great way to start thinking about approaching your voice in a more musical and rhythmic context.




2 > Zoom into your vocal, and hone in on a section that you think could be worked with. We've found a section in ours where we've got a little note sustain. Create a loop around a section and trim it out of the track.

3 > Next we'll take this vocal section, and open it in Logic's Quick Sampler. Here we can choose between assigning the sample in Classic Mode, which spans it across our MIDI keyboard range, or in One Shot mode, where any note will play the whole sample regardless of other notes played.

4 > You might notice popping or clicking from where the sample was trimmed out of the original vocal stem. It's often a good idea to adjust fade in and fade out points to smoothen out this harshness.

5 > Using Quick Sampler's synthesis and sound shaping controls, we can begin to manipulate the tone, filter settings and overall pitch of this now playable sound. An automated LFO, in sync with your project tempo, can really help to give the impression of movement.

6 > For more impactful stabs using a saturation plugin, such as Soundtoys Decapitator, can further energise the sound, while extra compression can help this new vocal instrument stick out in your mix. That's it! We hope we've armed you with enough tips and tutorials to help make your vocals work better in your music! **cm**

Download the accompanying video from this website: **filesilo.co.uk/bookazines**

# Mixing vocals with reverb

## Do your vocals sound washed out after reverb? Here are some tips for using reverb to add power to vox, rather than take it away

**> Reverb – it's an essential tool most people will be familiar with. It can add space, power, even emotion to vocals. You'd be hard pressed to find a modern pop song without it. Usually when new music makers come to adding reverb, there's a sound that they imagine in their head. But then... there's a completely different result that they get.**

A common mistake made by my new students, and even clients who send me a rough mix or their DAW sessions to mix and master, is to go crazy with plugins on every single channel. If there's a main vocal track and say two or three vocal backing tracks plus adlibs, there might be a reverb plugin on every single one of them!

Not only is this taking up valuable RAM and slowing down your session, it's also counterproductive in getting the desired sound. The different reverbs with different settings on each of the vocal tracks mean that the vocals are sitting in entirely different acoustic spaces and won't sound cohesive.

In putting reverb plugins directly on a vocal track, you're washing out the vocals, taking away the power from the performance the more you increase the effect. There is a better way to use reverb on your vocal tracks to maximise atmosphere while still maintaining the sharpness of the vox.

In this tutorial I'll share the most useful tips I've found through the years. From the correct use of reverb chains in your mixer to effective vocal stacking.

## >Step by step 1. The perfect balance

**1** > Setting up your session correctly for a new project is essential. I keep a template ready-made in Pro Tools to save me time when I start on a new project. But you can do this for any DAW. This template has two reverb auxiliary tracks. This means that the plugins are only on these tracks and *not* on the individual instrument tracks.

**2** > The reverbs on each auxiliary tracks have different settings. I will typically use one reverb channel for the vocals (so that all of the vocal tracks can be funnelled through a single chain and exist in the same acoustic space), and the other reverb channel for other instruments like drum components. Reverb tracks can be duplicated as needed, if you need more.

**3** > Duplicate the vocal tracks (and any other instrument tracks) that you will be applying the reverb to. This will double the volume of the vocals, so you'll need to turn the volume down on the duplicates in order to avoid having your ears blasted off. Come back to these duplicated tracks when you're ready to work on reverb later in the mix.

**4** > I add 'sends' to each duplicated vocal track that I will want reverb on. These sends make sure that the audio signal is sent through to the reverb auxiliary track of my choosing. For vocals, the destination track will be the same. But the key difference is how much signal I'm sending for each individual track.

**5** > Reverb, and other effects, should be applied towards the end of the mixing process. There are a number of key steps that you'll want to carry out first, such as the rough balance of your instruments, equalisation, compression etc. So I always deactivate the sends on the duplicate tracks, as well as keeping the volume faders down, till I'm ready to apply effects.

**6** > The backing vocals on this track I worked on were very heavy with reverb. The send fader is really high (right) compared to the send on the main vocal duplicate (left). A signal heavily processed with reverb is known as wet while an unprocessed vocal is known as dry.

## PRO TIP

### GETTING ACQUAINTED

Every DAW will have its own stock reverb plugins, and the standard nowadays is great compared to a decade or so ago. Before splashing out on third-party plugins, just get used to using all of the main controls and understanding what they do. Use a frequency analyst like Blue Cat's FreqAnalyst on your reverb auxiliary channel when you EQ verb on vocals and instruments. You'll see whereabouts the reverb effect is clogging up space in the frequency spectrum, and you can experiment with cutting frequencies to make your mix less muddy.

### SHOPPING AROUND

One of my favourite reverb plugins is PhoenixVerb by Exponential Audio (now owned by iZotope). I'd very much recommend signing up to iZotope's mailing list; they often have good offers. But you may also be familiar with pluginboutique.com – and if not, you should be! They've had a couple of absolute bargains on for PhoenixVerb this year. With this plugin there are a number of great presets to start with, but also a great level of in-built EQ control and processing thresholds. You can save anything you create as your own custom preset.

## Ashley Thorpe

**Ashley Thorpe (aka Breezewax) is a producer and musician who has created electronic music for over a decade. He typically uses a mix of samples, field recordings, synthesis and live instrumentation in his work, which covers a variety of styles including hip hop, chill hop, neo-soul and RnB. He also works as a music facilitator and freelance audio engineer.**

7 > As well as the send level, from each track to the reverb auxiliaries, there's also a balancing act to take into account in terms of the volume of the vocal tracks and their stacked duplicates. The volume of the original main vocal is higher here, and this means that the power of this original vocal is retained.

8 > I recommend turning the duplicate vocal fader right down and then bringing the volume back up until you have the right amount of reverb that you want. It's that simple, and means that you'll always have total control rather than just having a plugin slap bang on a track.

9 > Reverb settings can look daunting. Common controls are Dry/Wet faders (the amount of the original unaffected signal vs reverb amount); Attack; Decay (how long the reverb effect lasts on the sound); Size (aka room or shape, which affects the sound and space of the reverb effect); and Pre Delay (how long after the original sound the reverb starts).

10 > The size of your reverb is important. If you're going for an 80s power ballad style you'll want a large room effect. If you're going for an intimate, acoustic style you might want a small room, subtle effect just to make your vocals and instruments sound natural. In my example, the backing tracks are using large size with reverb Decay at an atmospheric 5.4 secs.

11 > Watch the amount of decay you add. Ensure it's fit for purpose for the style you're working in. If the reverb tail is too long and not EQ'd it can cause muddiness as the reverb tail clogs up acoustic space. For vocals you can definitely cut mid-range frequencies on your reverb; they won't add anything. Again, consider the most pleasant audible range.

12 > It's easy to go too far with reverb, and always be aware of the space you're listening to audio in. That, too, will affect how you hear things. This is where a good reference track comes in handy. Pick a song you know inside out, including how it sounds on different devices. **cm**

# MIXING DISTO

Give your tracks professional warmth, grunt and bite with our ten essential tips for turning up the heat

# WITH DISTORTION

**> Whether it's a subtle amount of tape saturation to add warmth to a clean synth part, retro bitcrushing to give a reverb send that crunchy 1980s digital feel, or over-the-top trashy distortion grunging up an entire drum kit, the various distortion plugins we have in our toolbox can open up new worlds of possibilities when mixing.**

Ironically, back in the day, mix engineers were always trying their hardest to avoid the characteristics of the hardware they were using, and dreamed of absolute sonic clarity; now, in the all-too-perfect software domain, we've since realised that in-the-box mixes can easily sound sterile and – dare we say it – *digital*. The best way to counteract this? With an expert application of saturation and distortion effects, of course!

These heat-inducing tools allow you to impart individual sounds, subgroups and mix busses with a near-endless range of colours and tones, and when applied properly, you can add the perfect amount of bite, character and energy to even the most vanilla of sounds.

So to help raise your distortion game to pro levels, we've assembled ten expert tips on how to give your signals and mixes a unique edge. Expect some killer info as we explore techniques including multiband distortion, multistage saturation, vintage tone emulation and even stereo applications. Grab our Tutorial Files, fire up your DAW... and turn up the heat!

## 01> Clipping peaks

Peak reduction is not only a transparent technique for mixing or mastering, but can also be an incredible creative tool when used correctly. Both limiting and clipping work by chopping off the loudest part of a signal's waveform – usually the transient – when said signal breaches a user-defined threshold.

As transients are of such a short duration, a moderate amount of peak reduction is fairly unnoticable, up to a point. In general, limiting is the more transparent process of the two, while clipping induces more noticable harmonic distortion – ideal for assertively flattening and fattening drum hits.

An advanced way to add punch to drums is to force those transients up into a stage of clipping. For example, add attack to a snare with an attack-boosting transient shaper, followed by a limiter or clipper. As you increase the attack, the limiter will stop the snare's loudening transient, increasing the perception of front-end punch while keeping its physical peak level the same.

## 02> Distorting ambience

Adding a splash of distortion before or after auxiliary return effects such as delay or reverb is a superb hack for giving those spatial effects some character. This works well for spicing up dub delay, as you distort the signal as the echo's feedback level rises.

A more subtle use for this is to place tape saturation before a drum reverb, giving the 'verb warmth and depth to fill out your drum mix in a way you wouldn't get by processing the drums directly.

## 03> Multiband heat

Multiband distortion plugins such as FabFilter's Saturn offer an endless variety of options for getting creative with distortion and saturation in the mix. Let's use Saturn to warm up a thin-sounding dance track...

**1 >** By splitting this track's bass across two Saturn bands, we can drive the sub and mid frequencies separately. Adding **Clean Tube** warmth and reducing the Dynamic range below **90Hz** beefs the sub up, while some aggressive **Power Amp** distortion with around **10%** of **Drive** gives the midrange more edge.

**2 >** To bulk up the midrange of the drums, call up three bands in a new Saturn, then **Bypass** bands **1** and **3**. After setting the crossover to **200Hz** and **2kHz** respectively, drive the second band with **Heavy Saturation** to add some weight without crunching up the rest of the bass's frequency content.

**3 >** Saturn's modulation lets us use **Smudge** distortion as a stuttering effect that sounds ace on this tune's intro pad. We're using an **LFO** in Slot 1 to modulate **Drive** amount, then automating the **Slot 1 Level** and **LFO Speed** over the first 24 bars to help create a build-up.

## 04> Heating up beats

There are few better tools for shaping the sound of a drum mix than saturation or distortion. If your percussion bus is sounding too spiky, for example, try applying some moderate tape saturation: the emulated effect of magnetic tape will naturally smooth harsh transients, giving the overall drum sound a less brittle, digital feel.

If you're after aggression instead, parallel distortion mixed on an aux return will give you the grunt you need, with the added advantage of being able to further process the distortion signal in isolation.

Plugins such as iZotope's Trash 2 or Cableguys' WaveShaper CM allow you to go even further and radically change drums with digital distortion or waveshaping. Ramming a drum through this kind of processing will instantly give you a tough, smashed sound that is superb for harder music styles. Although this can sometimes work on entire busses, you may bring out extreme artefacts that can spoil the effect – instead, for more control, process individual hits first, then edit out any overlapping artefacts afterwards.

## 05> Keep it tidy

It's surprising just how much high-frequency noise distortion can exacerbate or generate, even on low-pitched sounds like bass guitar. When mixing distorted guitars or synths, counteract this by dialling in a 1kHz high-shelf EQ cut to tame those upper harmonics – this'll give percussion and vocals more breathing room, resulting in a smoother mix.

## 06> Mono distortion

You'll often find that superficially impressive stereo parts such as synth leads or vocal acapellas will sound incredible when heard in stereo, but often fail to cut the mustard when listening in mono thanks to excessive phase cancellation between the left and right channels when they're summed into one.

In these scenarios, you can improve mono compatibility by sending the original signal to a mono aux return containing a distortion plugin. Not only does this make the sound stronger and beefier in mono, but it also doesn't affect the stereo imaging one bit. You can even automate the offending track's send volume to add emphasis at certain points in an arrangement – perhaps to make a naturally soft-sounding vocalist cut through for the chorus.

## 07> Retaining transients

Thanks to Dynamic Transient Control tech, Softube Harmonics adds grit when mixing, without squashing the sound's 'snap'

1 > First, group drum parts, then load Harmonics on the group. Choose the **Tube** drive type, then set **Drive** to **8**. The drums sound more gelled, but with dulled transients. Setting the **DTC Amount** to **8** restores lost snap, with **1dB Output** gain rebalancing volume.

2 > Some **Transformer** drive on our bass channel brings out the low-mid frequencies in a clean and clear manner. Setting **Drive** to **4** and **Character** to **7** adds some extra ruggedness and tone, with a **DTC Amount** of **2** bringing out the initial attack of each bass guitar note.

3 > Harmonics is an ideal tool for mix bus saturation. We put it on our master, and set it to **Master** mode. Increasing **Character** to **3** adds brightness, with a hard **Drive** at **8** gluing the mix together. A little **DTC** improves transient definition, with a **15kHz** low-pass filter tidying things up.

## 08> Vintage tones

Want to take your mixing back in time? Try using some age-appropriate processing! If you're after a classic 1980s pop sound, say, start your signal chain with some tape saturation followed by some API-or SSL-flavoured analogue emulation EQ and compression. This'll mimic an 80s analogue signal path, giving you a more convincing sound. With the multitude of plugins that emulate iconic studio gear from the 1940s onwards, a bit of research will give you all the information you need to create those retro sonics in the box.

## 09> Bass translation

Sometimes, bass that rocks on big studio monitors can get lost in translation when heard on smaller systems such as phones, tablets or Bluetooth speakers, resulting in a hollow-sounding mix. In this situation, you can add extra weight by routing a bit of bass signal to a distortion FX aux, followed by some high- and low-pass filtering to isolate the frequencies between 100-400Hz. This lets you mix in extra richness and depth to the mix's all-important low-mid frequencies, increasing the perception of bass weight on small speakers without distorting the low or high frequencies.

## 10> Chaining saturation

As with compressors and EQs, all saturation plugins differ in character – one tool may emulate a particular piece of outboard gear, while a clean, digital saturator may impart a more modern timbre.

Once you're familiar with the palette of drive flavours at your disposal, why not stack 'em up? Placing multiple saturation in series can give you a truly unique sound. For example, route one saturation stage into another, then automate the first plugin's drive amount to imbue an overly clean synth signal with custom grunt and movement.

Alternatively, try pairing classic tape drive with a digital-sounding saturator. Gentle tape saturation offers character without crushing the sound, which can then be pumped up by the cleaner plugin's crisp weight. **cm**

# MUSIC PRODUCER'S GUIDE TO SYNTHS

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# MIXING DRUMS

It can be difficult to pull all the elements of a drum kit into one cohesive mix. Here we have a step-by-step guide to processing and balancing along with some top drum mixing tips

**> Mixing drums is one of the most challenging aspects of music production. With so many variables, you can't just fly at it willy-nilly and hope to get good results. No, you have to know what you're doing, and here we're going to show you how to take a set of raw acoustic drum kit multitracks and mould them into a powerful, cohesive mix.**

Before you even touch those level faders, the one thing you absolutely must have is a decent set of multitracks, either recorded or from a quality ROMpler. No amount of mixing ability can make a rubbish recording sound good – unless you replace every single sound within it, which, although eminently possible, could have a detrimental effect on the overall feel of the groove.

So, if you're recording your drums, be sure that you know what you're doing in terms of mic selection and placement, room acoustics and drum kit setup and tuning. If you're using a ROMpler, it needs to be one that features separate virtual mic channels (preferably including separate top/bottom snare and in/out kick mics) and, ideally, control of mic bleed between channels.

So we're talking the likes of FXpansion BFD, Toontrack EZdrummer/Superior Drummer, NI Battery or XLN Audio Addictive Drums. If you can afford them, BFD 3 and Superior Drummer arguably lead the pack in this area, offering everything you need to make superb drum tracks.

We have two primary goals: achieving good separation between the elements of the kit, and imbuing our part with power, presence, character and 'that pro sound'. We'll do it mostly using EQ, gates and compressors – your three most valuable drum mixing tools. Incidentally, we're only dealing with backbeat-orientated styles here; jazz and gentler acoustic material require a looser, broad-dynamic approach.

# Drum miking 101

To mix drums, you need to understand miking techniques. There are two fundamental ways of miking up a drum kit: close and ambient.

Close mics are positioned to capture the direct, 'isolated' sound from one drum or cymbal in the kit. The other drums will inevitably be picked up in the background – this is called bleed or spill. While it usually needs to be reduced with EQ and gating, you don't necessarily want to eradicate it entirely, because a bit of bleed makes the kit sound more natural and cohesive.

Toms and cymbals are almost always single-miked. One mic inside the kick drum is sufficient, but some engineers also use an outside kick mic and maybe a special 'sub' mic to capture the low end. These can be blended to taste in the mix, provided they're in phase (which we'll come on to shortly).

The snare drum most often uses two mics: one on the top head and another underneath, pointed at the bottom head. The latter captures the sound of the snare wires rattling, while the former captures the 'thwack' of the stick striking skin.

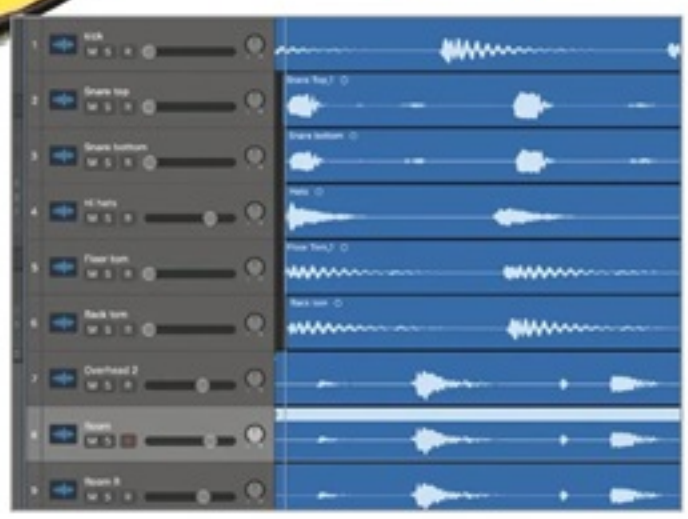
When mixing, you can process the top and bottom mics to achieve the desired balance of oomph and sizzle.

Ambient mics capture the sound of the whole kit from a particular perspective. Overhead mics are the most commonly used – typically a stereo pair placed above the kit. You may also encounter room mics, which are placed further from the kit to capture more of the recording space's natural reverb.

Turning all these mics up at once will lead to undesirable interactions between the channels. This can be reduced using gates (to silence the signal between each hit) and EQ (to remove frequencies that belong to other drums/mics) on individual channels.

The fundamental (lowest) tones are particularly important in conveying power and punch, so high-pass filters and EQ can be handy to ensure that the fundamental tones of the kick and snare are supplied mostly by their internal and top mics respectively.

## > Step by step Mixing live drums



1 > Although we're working with a set of live recordings, everything we're about to do applies equally to a virtual drum kit plug-in. Such plug-ins will ideally give you total control over mic bleed – a definite plus. So, here are our nine tracks/channels: Kick, Snare (Top and Bottom), Hi-hats, Floor Tom, Rack Tom, Overheads and L/R Room.

2 > There's a bit of spill from the snare and toms in the Kick channel, so we insert a gate plug-in to clean it up. We set the Threshold to -16dB in order to let only the kick through, the Attack and Hold times to 0ms, the Lookahead to 6.8ms, and the Release to 310ms – long enough to let the full decay of the drum ring out. We do this with the Kick channel soloed, as no context is required for purely corrective work.

3 > Let's up the kick's power and definition. We EQ first, activating a low-cut filter at 30Hz to remove headroom-eating ultra-lows, boosting by a hefty 6.5dB at 70Hz for welly, boosting 5.5dB at 5.4kHz for 'click', and cutting 9dB at 152Hz and 445Hz to reduce boxiness and make space. Our compressor is set to catch every hit at a 9:1 Ratio, with a long enough Attack to let the transient through.

POWER TIP

### >Phasing issues

If your top and bottom snare drum channels sound strangely hollow and, well... weird together, chances are that they're out of phase due to the polarity of one of the mics being inverted. Fixing this is a simple matter of loading the requisite utility plugin in your DAW and hitting the **Phase Reverse** or **Invert** switch. Some DAWs even have this button built into the mixer. The snare in our track is fine, though, so we don't need to do this.

4 > There's a lot of sympathetic resonance coming from the Rack Tom channel, and we need to gate out the bleed into the Floor Tom mic, too. We load gates onto each channel, both set to -9dB Threshold, with Release times of 351ms for Rack and 414ms for Floor. There's still a bit of bleed at the end of the Floor Tom hit, but there's no way to avoid that if we're wanting to keep the sustain.

5 > We tackle our Snare Top and Bottom channels together. Again, there's plenty of spill into both mics from the other channels, so we set up a pair of gates to do away with as much of it as possible without killing any ghost notes. This is only mildly effective, but it makes enough of a difference to the track to warrant doing, drying out the Bottom Snare track in particular.

6 > Now for some EQ. We high-pass the Top and Bottom channels with the steepest filter slope (48dB/oct) at 70Hz (for Top) and 180Hz (Bottom). Boosts at 200Hz (2.5dB), 1.4kHz (4.5dB) and 8.3kHz (3.5dB) give the Top channel body and snap, while deep, tight cuts at 670 and 970Hz do away with any odd overtones. A compressor with a hefty 19:1 Ratio on the Top adds oomph. EQ the Bottom as shown.

7 > The gating and other processing has dried the snare out, so we load a small plate reverb onto an auxiliary channel and send a low amount of both Snare channels to it. The Kick channel spill gets caught up in this in a good way, putting a bit of tail back on our otherwise rather dry kick drum. With the send levels low, the effect is subtle – you don't really notice this 0.7-second reverb until it's bypassed.

POWER TIP

## > To pan or not to pan?

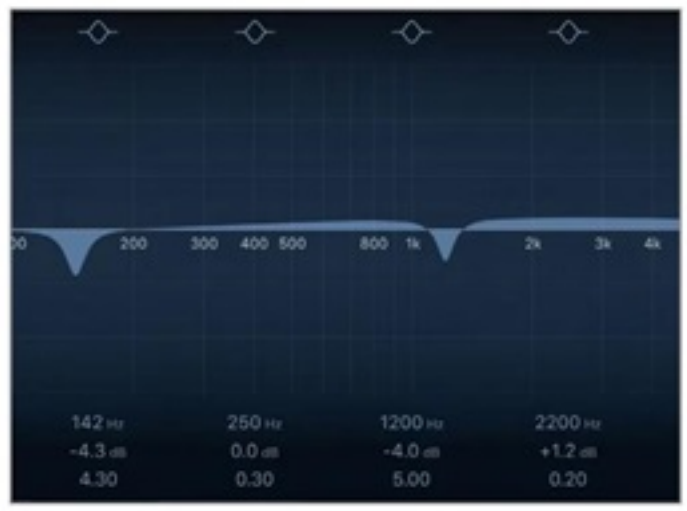
Generally, you want your kit to sit centrally, especially the kick and snare. Unless you're going for an odd effect or a very retro sound, leave those two dead centre. Hi-hats can sound good panned slightly left or right, however, and toms can also be panned to descend/ascend across the soundstage – just don't go too wide. Overheads will usually be in stereo already (or on two mono channels), but you might want to widen or narrow them further, depending on your track.

8 > The hi-hats are the easiest part of the kit to deal with. In this case, all we have to do is apply a high-pass EQ to attenuate everything below 290Hz – spill from the kick and snare – and then add a gate to get rid of most of said spill completely. We also throw in a 6.5dB EQ boost at 14kHz to up the brightness and 'air', bring the level down a bit, pan the hats left slightly, and dial in a low send to the snare reverb.

9 > Our two Tom Tom channels have already been gated to silence for the vast majority of the track, doing away with unwanted resonance and spill. They need real impact when they arrive, though, so we give them EQ boosts in the low mids and highs, for stick impact and body, and moderately hard compression for greater presence in the mix. We pan them left and right, and send them lightly to the reverb.

10 > With our direct mics done, all that remains are the overheads and room channels. The overheads are crucial to a good drum mix – they enhance pretty much every element of the kit, particularly the snare and cymbals. Here the overheads are of most importance in the second half, when the drummer takes to the cymbals. We apply 5.5dB of high-shelving boost from 4.4kHz, and turn the level down a bit.

11 > The overheads have added a good amount of width and sheen, but we're lucky enough to have left and right room channels, too, for natural reverb. We pan these left and right, filtering off everything below 150Hz to keep the bottom end tight. Getting the level balance correct is crucial with room channels – dial in too much and your kit will sound unfocused.




12 > Finally, we bring it all together with a group EQ (moderate, tight cuts at 140Hz and 1.2kHz, and a small boost at 2.2kHz) and a tube levelling amplifier plugin. We don't go overboard with the Gain Reduction, since we're not looking to dramatically crush our drums, and set a roughly 50/50 mix between wet and dry.

POWER TIP

## > In a full mix

We're mixing our drums entirely for their own sake in this tutorial, but the next step would be to get them sitting perfectly within the context of a full band. This would probably involve tweaking the overheads and any reverb we might have set up, nudging the kick drum EQ to fit it around the bass, and generally finessing the drums while being sure to maintain their cohesion and authority. Throughout the drums mixing process, bring the other instruments up from time to time, as well as giving the kit plenty of solo attention.

# Drum mixing tips

## REVERB INSERTS

**Although we used a single bussed reverb in our walkthrough, there's no reason not to use inserted reverbs on your separate channels. A short, dark 'verb on the kick, a longer, brighter one on the snare, etc. If you choose to do this, though, be careful not to make it sound like each of your drums is in a different space to the others – unless you're going for deliberate weirdness, that is.**

## PARALLEL COMPRESSION

**Parallel or 'New York-style' compression is simply the act of mixing compressed and uncompressed versions of the same signal in order to reduce the dynamic range but retain all those punchy attack transients. Either bus your drum group through an auxiliary track with a compressor plugin inserted and mix it with the original, or use a compressor with a wet/dry mix control as we did earlier.**

## SIDECHAIN THE OVERHEADS

**A range of cool effects can be created by putting a compressor/gate on the overheads channel and keying it from the kick, snare or hi-hats. Key a frequency-conscious gate from the snare to emphasise the airy overhead signal with each snare hit, or compress the overheads via the kick to draw the sides in with each kick hit.**

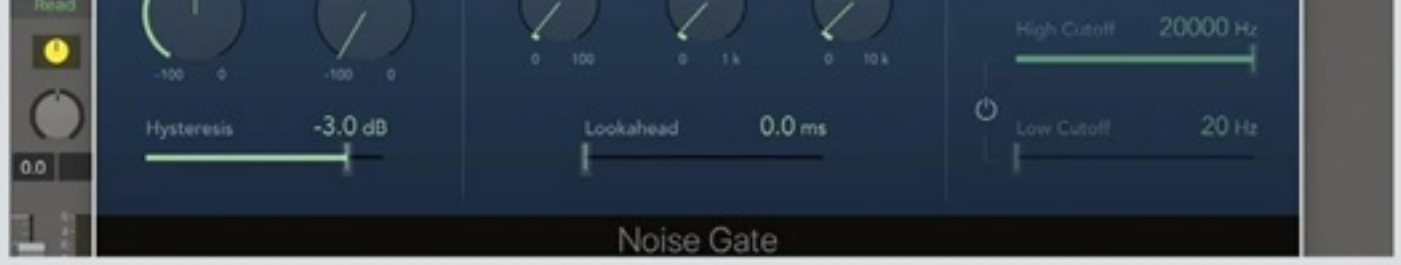
## OVERHEAD PROCESSING

**Applying a stereo widening plugin to the overheads channel can work wonders for the sense of breadth. Two things to avoid with overheads, though, are overly heavy EQ boosts above 10kHz and compression (though a touch of compression can be effective for a vintage-style sound). And if your overheads sound unexpectedly weird in any sense alongside the close mic channels, don't forget to try flipping the phase.**

## BRING THE NOISE

**If the wires on your snare drum aren't making their presence felt strongly enough, white noise can be used to add a synthesised back-up layer. Load any analogue synth with just a white noise oscillator active and tune it quite high. Draw in a single MIDI note for the duration of your drum track and insert a sidechainable gate plugin onto the synth's mixer channel. Key the sidechain from your snare track and set the gate threshold and envelope parameters so that the synth only triggers when the snare hits, then fades out quickly and smoothly.**

**Keep the level of the fake 'wires' low in the mix, and be aware that it might need muting during fills. The same trick can be used to add sheen to hi-hats and even – using a sine wave – sub-bass to kick drums.**




**If you need to give your snare some extra bite, some cunning white noise oscillator layering can really help**

## JUST A PHASE

**Because a drum hit will take slightly longer to reach the overhead mics than it does the drum's own close mics, its timing will be slightly smeared across the channels. This is part of the sound of recorded drums, and it can help to create some cool tones as well as smoothen sharp transients. However, too much can sap punch, power and clarity.**

**You may be able to manually adjust the timing of the recordings in order to improve their combined sound. In our example audio files, a subtle increase in low-end solidity can be created by moving the snare bottom mic channel back into time by about 4ms, so that the 'meat' of the waveform becomes a little bit more neatly aligned with that of the snare top mic channel.**

**Remember that you can typically only align one drum (the snare is your best bet), as the timing offsets will vary due to each piece of the kit being at different distances from the mics. And don't get too obsessed with aligning your tracks – they might sound better messy!**

## TAKE THE BUS

**In our walkthrough we processed every mic channel separately, but combining related drums/cymbals in group busses can be both convenient and sonically beneficial. For example, grouping all your toms on one bus and inserting a compressor, an EQ and a reverb into that, rather than giving each separate tom its own treatment, can make them sound more cohesive, as well as placing them all on one handy level fader and saving on CPU cycles. Bargain!**

## SAMPLE MAGIC

**While the idea when mixing a drum kit is to make the results sound like the bringing together of a set of simultaneously played elements to create a cohesive single instrument, there's no point making life unnecessarily difficult for yourself in the name of keeping everything organic. If your drummer neglected to hit an essential crash cymbal at the end of a key fill, don't drive yourself mad trying to paste in a cymbal from elsewhere in the overheads track – just throw in a crash from your sample library (on a separate track, otherwise it'll sound like a dodgy punch-in). Similarly, if your snare's just lacking a bit of top but is otherwise OK, double it up with a high-passed sample. Play with the mix settings, A/B with some well-chosen reference material and trust your ears more than your eyes.**

## SINGLE HITS

**Further to the previous tip, when recording your drums, be sure to capture your drummer hitting each piece of the kit on its own a few times at hard, medium and soft stroke strengths. Then, if you do need to layer in any 'overdubs' (or even replace any sounds completely), you'll have a set of perfectly matched sounds with which to do it. cm**

**You can use separate reverbs on each drum channel if you like, but be careful that they're not too different!**

Get the tutorial files on your PC/Mac at **filesilo.co.uk/bookazines**

# Mixing your kick and bass

Getting kick and bass to complement each other can be one of the hardest production tasks. Follow our guide to master your low end

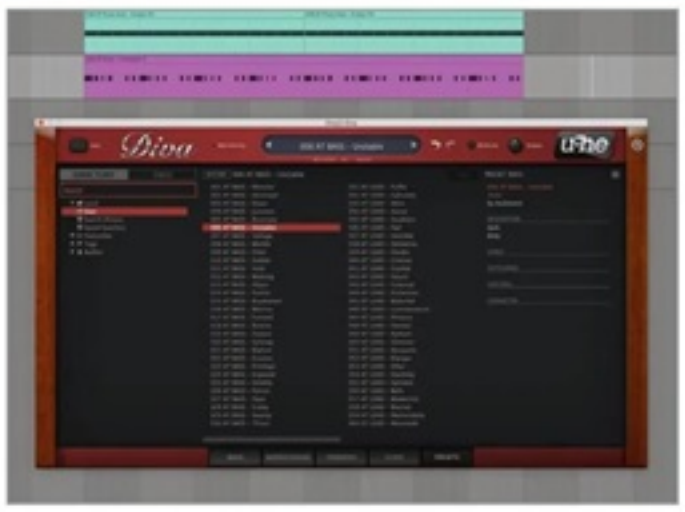
> Mixing low end is one of the most requested topics to cover: a lot of producers struggle with it, mostly because the room acoustics and speaker setup isn't ideal. But there are ways to address overlapping frequencies, and here we'll use our favourite, tried and tested mixing techniques to get your low end thick and powerful. We'll start with the obvious – but often overlooked – level balancing and tuning of the bass. As always, fixing the sound at the source brings the best results. Therefore we first try to use tricks like shortening the release of the bass within the synth, before going extensively into processing. Shortening the release can make a huge difference as it tightens the bass and stops the decay conflicting with the kick drum.

If you don't have full control over the synth parameters and are using an audio sample as a source, more traditional processes can be used to solve the problem. Sometimes a simple complementary EQ is all you need. If that doesn't help, we can further sculpt the balance of your low end by applying a frequency dependent sidechain. This not only helps to make vital space for the kick, but also leaves the upper harmonics of the bass intact.

The low end is the foundation of dance genres and it's worth dedicating studio hours to it. Get this key element right and everything else will fall into place a lot easier.

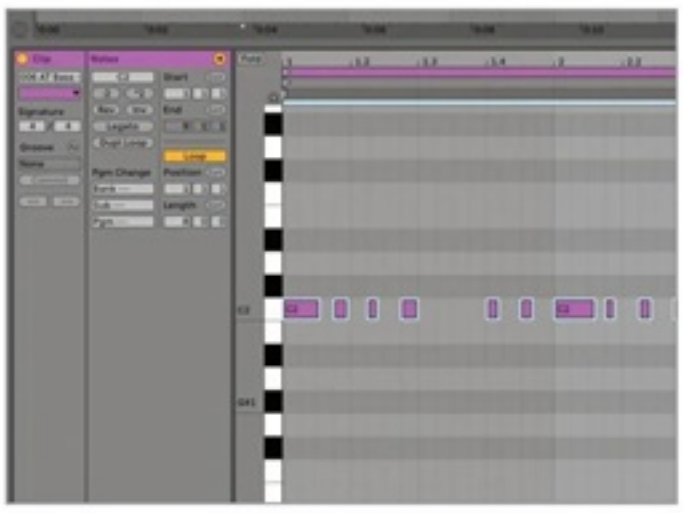
## >Step by step Clearing out your low end

**1** > To set the foundation for the track, import a kick drum into Ableton's MIDI Simpler device. You could also insert it directly into the arrangement as an audio file. We are using one from Audiotent's Pure Kicks library. These are all sourced from analogue drum machines and Eurorack modules. A clean sample is preferred to an overly processed one.

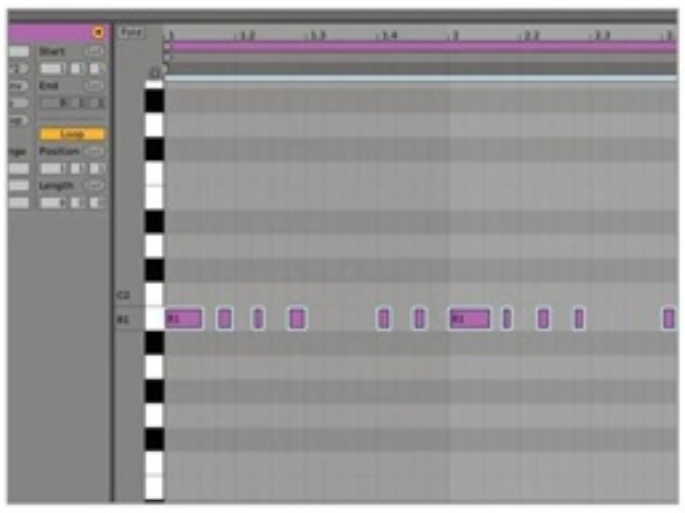



**2** > Now, let's add a synth bass. One of our favourite instruments for the task is u-he Diva. Not only that but (plug alert!) Audiotent produce preset libraries for it so we've chosen one from our very own Rectify library. It sounds quite good already. However, the bass overlaps with the kick in a few places, so there is some work to do.

**3** > The first and most important process of getting your kick and bass to play nicely together is to find the right balance between them. Generally, in the electronic dance music genres, we prefer to prioritise the kick drum. Bass is mixed slightly below in volume.



**4** > One of the most effective ways to make kick and bass work together is tuning one of them till they work as a team. If possible, it's best to alter the MIDI info of the bass synth, rather than repitching an audiofile – some pitch algorithms may introduce undesirable artefacts.



**5** > Here, we've pitched the bass up and down a few semitones to find a sweet spot in relation to the kick. Normally, if the kick is longer and has a clear tone, you'd tune the bass until the kick sits in a scale. In our case, we pitched the bass one semitone down from C2 to B1 as we plan to use a B minor scale. The kick is tuned to F♯ a perfect fifth in relation to B minor bass.

**6** > As we said, it's always best to fix problems at the very source if possible. Since we have access to the original bass synth patch, we can try shortening the release part of the sound. This will prevent the decay of the bass interfering with the tail of the kick.

## PRO TIPS

### BASS IMAGING

Keeping your bass in mono is vital if you want to achieve a solid mix. However, sometimes you might want a little extra width in your bass to fill out the panorama. Simply duplicate your bass and high pass around 250-350Hz. Next, choose your favourite imager, chorus or Haas effect plugin to increase the width of the bass. The result is a full-sounding low end while retaining the mono image in the subs.

### BASS EXCITER

If your bass feels slightly muddy or undefined in the mix, why not try adding a subtle exciter in order to enhance the upper harmonics? This technique can often bring the bass to a more upfront position and give an extra bit of bite to help it cut through the track. Watch out for any undesired distortion in the lowest frequencies as too many harmonics in that area can also make the bass sound muddy.

**Use an exciter plugin (subtly) for more upfront bass**

## Ed Strazdas

**Ed is an experienced dance producer and sound designer, working in the scene since 2004**

## Neil Crockett

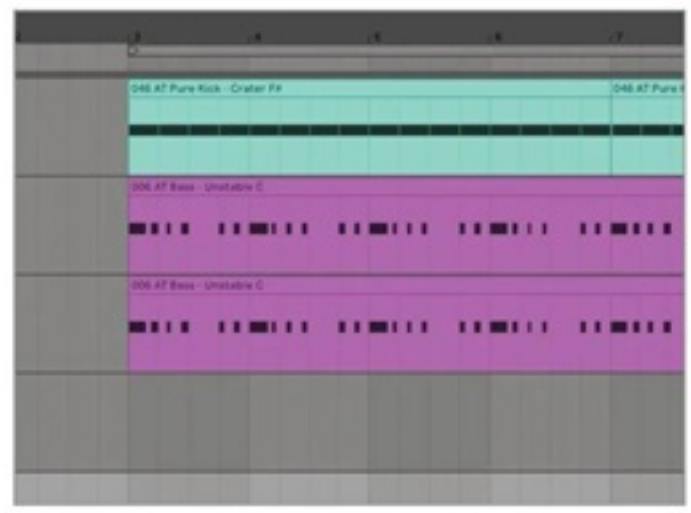
**Sound designer and mixing engineer, Neil is a huge fan of Eurorack modular synths**


## About Audiotent

Our Production Technique experts are from Audiotent, a team of creative producers and sound designers who provide inspirational tools and education for professional artists worldwide. They focus on both techno and progressive genres, offering presets, samples, templates, MIDI files and production tutorials. More from: **audiotent.com**

**7** > Once you achieved a nice level and timing between the two elements it could be further refined with EQ. The kick seems to have a slight frequency build up around 260Hz. A 4.5dB cut at 264Hz was made to make space for the upper bass harmonics. A gentle 2.3dB boost at 46Hz further extends the fundamental frequency of the kick.

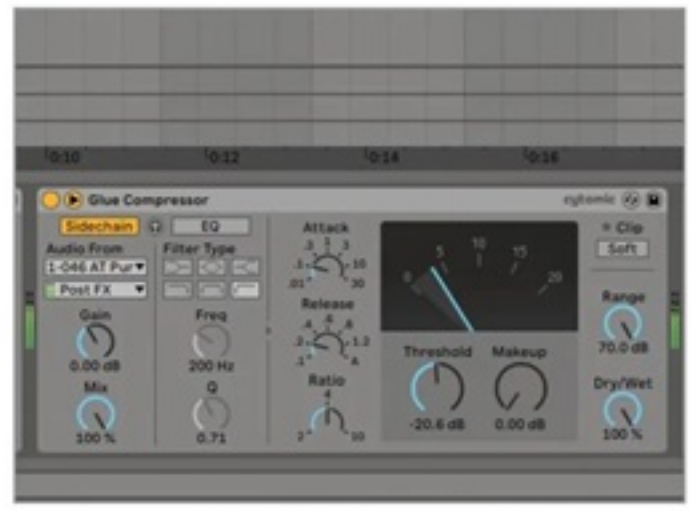
**8** > Next, we follow the similar procedure for the bass. A high-pass filter is applied at 46Hz, rolling of the lowest unnecessary frequencies. This is a fairly high frequency to cut at, but since we don't have any notes lower than B1 for the bass, it's OK to cut here. Everything that's below this frequency is eating into our precious headroom.



**9** > As a final process, a sidechain trick can be used to give room back to the kick. We won't sidechain the higher frequencies of the bass, with that obvious ducking effect. To achieve this, only the sub frequency of the bass will be sidechained. Therefore we'll have to duplicate the bass channel and use EQ to split them into separate frequency bands.

**10** > Now we have two bass channels, the EQ settings can be adjusted on each to isolate the sub. First, for the 'sub' bass, low-pass filter at 150Hz is applied, cutting the upper harmonics of the bass away. Second, the 'high' bass can have it's high-pass filter raised to 150Hz. We then get a nice crossover point that won't intrude on the fundamental frequency of the bass.

**11** > The sub frequency is quite dynamic and could benefit from a limiter to control its range of motion. 2 or 3dB maximum range of limiting is being applied, which seems like a nice sweet spot and sounds natural and controlled. The output volume is level-matched, so we are not tricked that louder is better.




**12** > Finally, apply the sidechain compression to the sub layer of the bass. Ableton's Glue Compressor is perfect, due to its internal sidechain routing. With the compressor inserted, choose the kick as a trigger. Short attack and release settings help the compressor recoup quickly and cleanly. Threshold is lowered until we get the desired effect. **cm**

# GET DYNAMIC

## Dynamic processing is one of the most useful – and misunderstood – parts of mixing and production. Here's how to master this most important and creative of processes, forever!

**> Music would be very boring indeed without variation, and that's exactly what musical dynamics are all about: variations in loudness. Examples might be a quiet verse leading into a loud chorus, a synth line that gradually increases in volume, some notes within a pattern played louder to accent them, a singer belting out certain words for emphasis, and so on. There may be unintended variations too, such as a bassist or drummer unable to play each note or hit at a constant volume where required.**

Fixing unintentional problems is one thing, but even deliberate accents and changes in volume will almost certainly need some control to fit the norms of what we as listeners expect to hear in a finished, cohesive mixdown, free of unexpected jolts in level and excessive variance. While mixing, you may also discover a build-up of low-level sounds such as background noise – too much of this can pile up, resulting in a cluttered, unclear mix.

So what can we do about all this? How can we get dynamics under control to create a solid, professional mix? Well, you could use volume automation to briefly turn down any overly loud notes, to cut out background noise, or to bring up the level where required, and so on... But that'd be a right old hassle. There's got to be an easier way, right?

You're in luck: help is at hand in the form of the many dynamics processors available to the modern producer, such as compressors, expanders, noise gates, transient shapers and limiters. These effects take the hard work out of managing levels, and in fact, they are more than mere functional tools, offering the possibility of imparting such qualities as character, 'glue', movement, punch and snap to your tracks when used wisely. For even greater precision, power and flexibility, there's frequency-conscious compression, multiband dynamics and parallel processing.

Now the bad news: Dynamics processors are not especially intuitive, and many producers struggle to understand how they really work, which sometimes results in them doing more harm than good. What you need, then, is a solid grounding in the fundamentals of how these vital mix tools work, of the scientific-sounding terminology that goes along with them, and the tried-and-tested techniques and starting points that'll allow you to use them with confidence. And – wouldn't you know it? That's exactly what we've prepared for you right here! Now fire up your DAW, open your plugin folder and get ready to *Get Dynamic* once and for all...

# Introducing dynamics processors

When we talk about adding, reducing, changing or enhancing dynamics, we are, in essence, talking about manipulating volume levels. Dynamics exist at both a macro and micro scale, with the former concerning overall track levels and the latter the dynamics specific to a sound's own amplitude envelope. The key tools we have for macro dynamics include broad-brush techniques such as song instrumentation, fader level rides and simply muting elements at certain times. At the micro scale, we *can* use these same techniques, but the vast majority of micro dynamics are handled by a set of tools called dynamics processors.

To understand what a dynamics processor is and how it works, let's look again at how a sound's levels change over time. Typically, it will have an attack stage, a sustained stage and a release. As an example, a piano note may have a sudden loud attack, preceded by a fast fade down to half the attack's maximum level, then a much longer tail fading out to silence over time.

So, there are differences in level between a sound's loudest and quietest parts, and we may want to change those differences, enhancing the quiet parts, lowering the louder parts or removing the quietest parts altogether.

Dynamics processors are level-dependent effects that allow us to do this in real time, based on the incoming signal level and parameters we set on them. Typical tools are compressors, limiters, gates and expanders, but we can also add to the list de-essers, duckers, 'enhancers', frequency-conscious compressors and multiband dynamics processors. There are also transient shapers that impose a user-adjustable envelope, triggered by the transient of a sound.

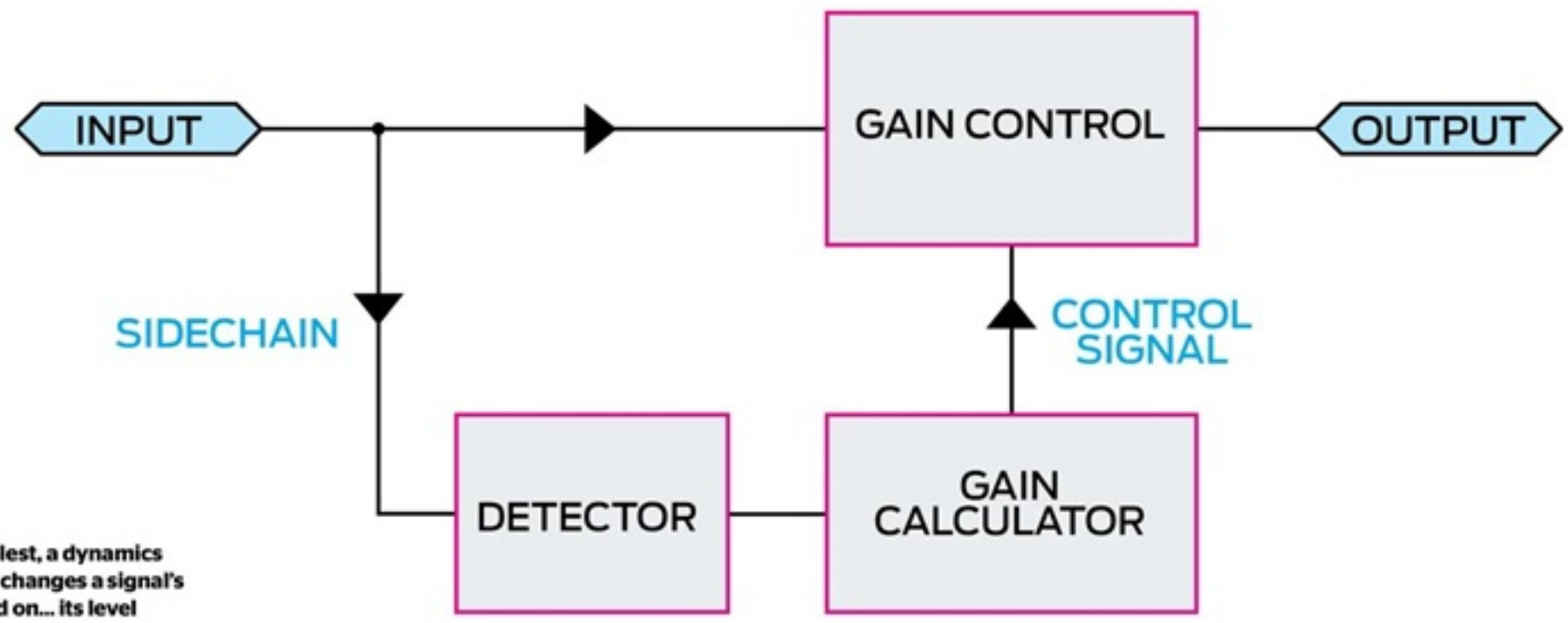
## Making a good compression

So how do these devices work under the hood? Before we talk tech, imagine this simple – if slightly far-fetched – scenario: inside your compressor is a tiny imp with its hand on a single, plain-old volume knob that can be used to adjust the outgoing signal's level. The imp's job is to constantly monitor the level of the incoming audio signal, and if it exceeds a certain level – called the threshold – to spring into action, turning down the volume knob by a proportional amount, known as the ratio. The more the input signal exceeds the threshold, the more the imp must reduce the output volume.

There's more, though, as the poor imp has been given exact instructions as to the amount of time it should take to turn the volume knob down when this happens, whether it's a lightning-fast fraction of a millisecond, or a leisurely half a second... or something in between. This is known as the attack time. That's not all, either: what goes up must come down, and once the incoming signal level backs off, our beleaguered imp has to bring the volume level back up accordingly, again following a precise instruction as to how long it should take, known as the release time.

That is, more or less, how a compressor works, though in the real world you'd have to replace the imp with a chunk of code to continuously track the input level (known as the 'detector'), calculate how the output level should be modified (the 'gain calculator'), and then control the final output volume accordingly via a gain control. You can see in our diagram below how these core components interact.

But what of the many other types of dynamics processors? They are, for the most part, variations on the above theme, just with slightly different instructions for our theoretical imp. For instance, in an expander, the imp would be charged with *reducing* the volume whenever it falls *below* a certain level.




**At its simplest, a dynamics processor changes a signal's level based on... its level**

## Functionality versus creativity

Dynamics processors started out as purely functional devices, and in fact we can trace compressors (or leveling amplifiers as they were called) back to the early days of radio, when they were used to protect transmitters from overmodulation (and likely failure). Audio compressors in music engineering only became popular in the very late 1950s with the likes of the Teletronix LA series and Fairchild's 670. Around this time, more complex dynamics processors were also finding their way into vinyl cutting, and so-called acceleration limiters (high-frequency de-esser style limiters) were often used during the cutting process.

Today we still use dynamics processors for functional purposes, including compressors as gain-leveling devices, limiters to curtail peaks and prevent overloads, de-essers for softening vocal sibilance, gates/expanders for removing or reducing background noise from recorded tracks, and multiband dynamics to keep our level in check on the mix buss.

However, useful though these are, it's the creative applications that tend to draw the headlines. One of the most common uses for a compressor is to increase an instrument's sustain, and you can easily take this technique one stage further to enhance the snappiness of shorter sounds such as drum hits. Push a compressor a bit further with a slower attack and fast release and you can start to make things 'pump' audibly. Add in a rhythmic external sidechain, such as a kick drum, and sounds can pump in time with the beat itself. That's quite a few ideas from one simple level-based processor.

Furthermore, you can fine-tune the attack and release stages of the compressor to shape the envelope of a sound. Or try the same thing with a gate or transient shaper. And finally, we could rig up an external sidechain for our gate and use it to create cool rhythmic effects.

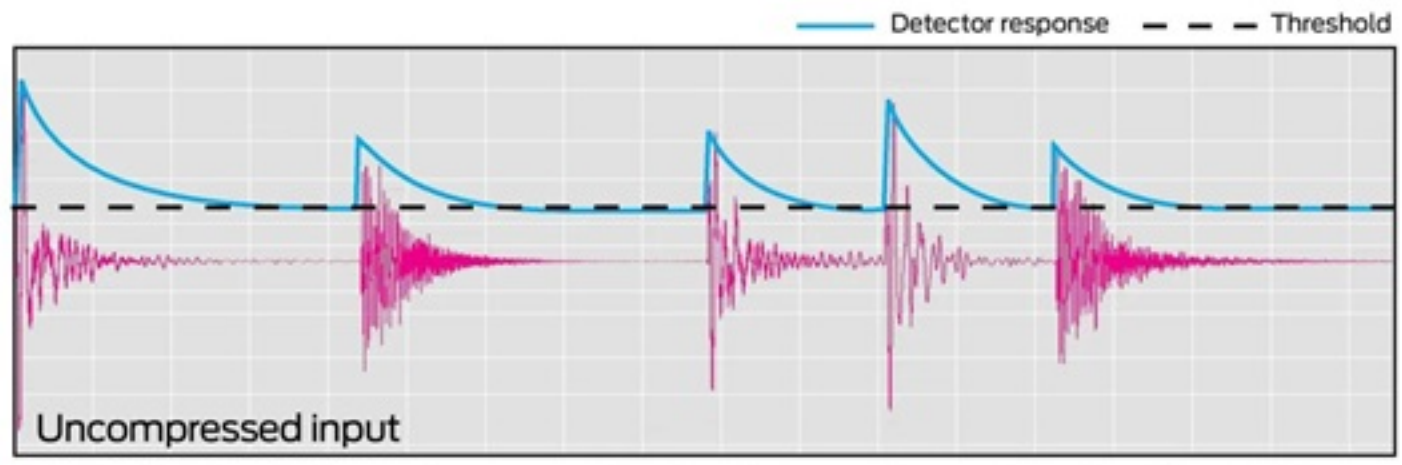
# Compression

Audio or 'dynamic range' compression is a level-dependent process that reduces the dynamic range of a signal (makes loud bits quieter). When the signal level exceeds a certain **threshold**, the gain reduction kicks in.

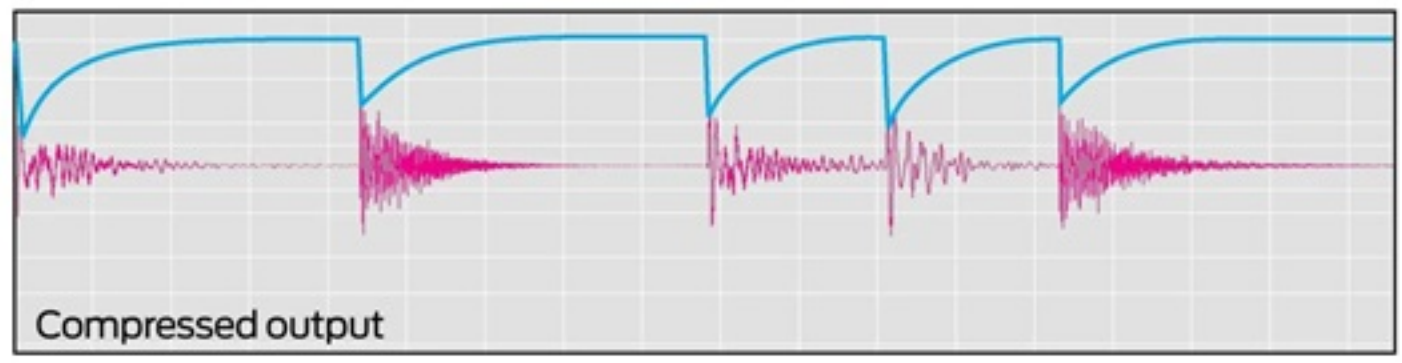
The amount of reduction is determined by a **ratio** setting. A setting of 2:1, for example, would reduce the level of any signal above the threshold to half the amount above the threshold. Exceed your threshold by 2dB, and the signal would be reduced by 1dB; exceed it by 6dB, and the signal would be reduced by 3dB.

This does not happen instantaneously, as the compressor take a little time to make this change in volume once detected, set by the user with the **attack** control. Likewise, the user has control over the **release** time. This has a big impact on how the **gain reduction** sounds (some compressors have a meter that shows the amount of gain reduction (GR) taking place).

The basic compression concept is easily demonstrated on our transfer graph below, where you'll see that the 4:1 line above the threshold is shallower, with a slope based on the compression ratio. This method is known as '**downwards**' compression, and is by far the most common type. Certain designs reduce dynamic range by increasing the level below a threshold. This is called '**upwards**' compression and is not only much rarer, but also harder to fine-tune. Since the vast majority of compressors are downwards types, that's what we'll focus on.




**1. The blue line shows the detector's response to a drum beat. In our example, as each new drum hit exceeds the threshold, the blue detector line jumps up quickly (fast attack), then falls off gradually (moderate release)**




**2. Here, we've flipped the response curve upside down to help you visualise how the detector's output is used to 'push down' the signal level. The attack of each hit is now quieter, but the tail is the same level – that's compression!**

## Further controls

So then, compression reduces dynamic range, but it doesn't have to make the signal quieter. This is because compressors have a so-called "**make up**" stage (either manual or automatic) to bring the overall output level back up to the original input level (or higher or lower).

There are further factors that influence how a compressor behaves. Firstly, **knee** shape: This refers to how gain reduction is introduced as the signal approaches the threshold, and can be abrupt (**hard knee**) or gentle (**soft knee**) or somewhere in between. See our graph below right for an example of soft vs hard knee.

The way the detector senses the signal can vary and two typical methods are **peak** and **RMS** (root mean squared) sensing. The former responds precisely to level changes, while the latter reacts to the average level, meaning less of a response to abrupt events like transients.

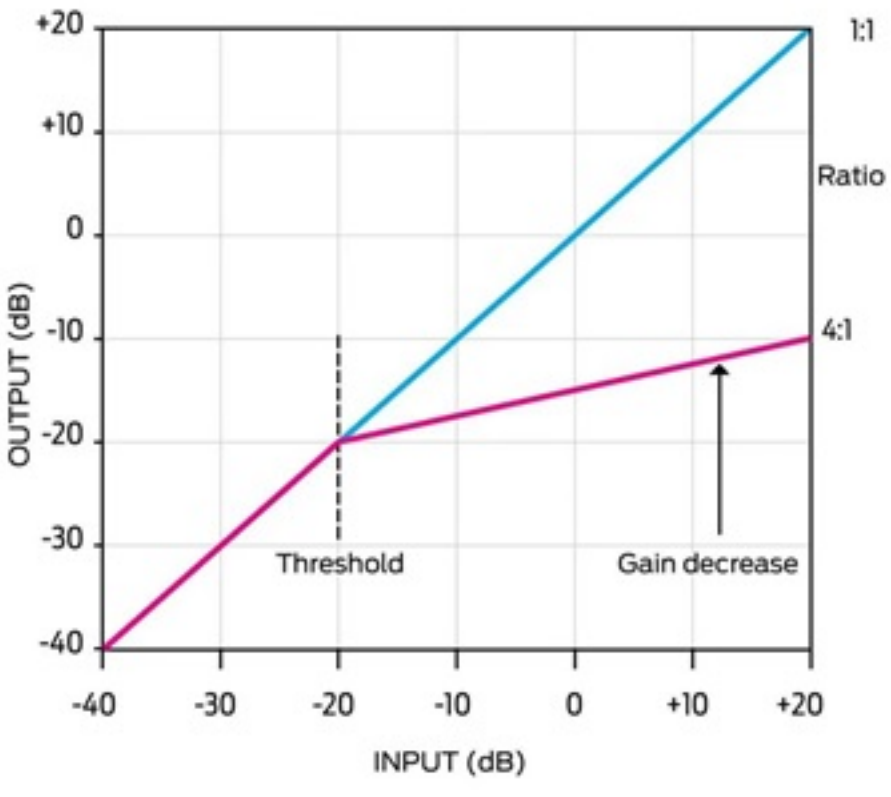
Next up, the **internal sidechain**, which is the signal that feeds the detector. In most modern designs, this is split off from the input signal, known as a **feed-forward** topology, for a precise response. Many vintage designs, however, are of a **feedback** design, where the sidechain comes from the (possibly already compressed) output signal, giving a less predictable result.

In our examples so far, the sidechain signal has been untampered with, but a compressor may have EQ or filtering applied here, known as the **sidechain filter**. For example, high-pass filtering the sidechain signal means that the detector 'ignores' bass frequencies, and they do not trigger the compression. Note that this filtering is applied only to the sidechain signal and not the 'pass through' signal that we hear.

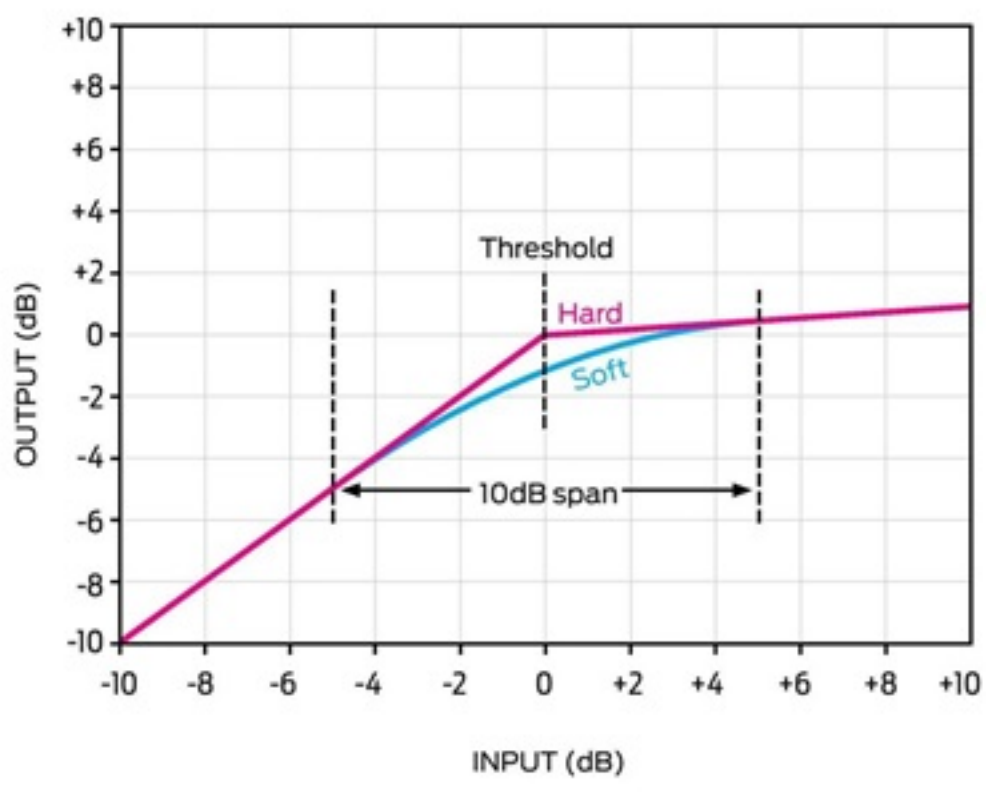
Also look out for **stereo linking**. This uses a combined left/right signal for the sidechain, to keep the stereo image solid and centred.

The **lookahead** option allows a dynamics device to see what's coming up in the signal and react to it ahead of time, for a smoother sound and the ability to catch very fast transients. Plugins achieve this by delaying the signal 'under the hood' before it hits the gain reduction stage, but sending the undelayed version to the detector, so the compressor reacts to things before they are heard. The resulting delay/latency should be compensated for by your DAW.

Traditionally, a compressor with a high ratio (10:1 upwards) was called a **limiter**, and with a very high ratio (30:1 upwards), a **brickwall limiter**. However, in the modern digital era, unless stated otherwise, limiter plugins are almost all of the **digital brickwall** variety, not allowing any signal to exceed the threshold.




**Transfer graph of 1:1 (blue) and 4:1 (pink) compression**




**The soft-knee curve here has a 10dB span, though this is purely for example**

## > Step by step 1. Exploring basic compression parameters

**1** > With a drum part loaded (there's one in Tutorial Files), add a compressor that has a gain reduction meter – we're using CM-COMP 87 from **cm** Plugins. If your plugin has automatic makeup gain, turn it off, and set the Ratio to 4:1. Starting with the Threshold at 0dB, gradually reduce it until you start to see a few dB of gain reduction on the Reduc meter. Note how the output signal level is reduced.

**2** > Set Attack to 50ms, Release to about 100ms, and adjust the Threshold to achieve moderate gain reduction on the Reduc meter – go for -5dB for starters. Gradually lowering the Attack, as shown, increasingly compresses the transients; lengthening it lets transients through and brings up the body of the sound. Think of the Attack as effectively adjusting the duration of transients. Set it back to 30ms.

**3** > Raise the Threshold by 3-4dB and gradually reduce the Release time. As the Release gets shorter, the sound begins to pump more and the transients become more punchy. If you're not hearing it, lower the Threshold to emphasise the effect. Revisit the Attack control and try both slower and faster settings. Listen to how the Attack and Release work together to shape the compression character.

**4** > Set the Threshold to -25dB, Attack to around 20ms and Release to 50ms. Now switch through a few Ratio settings. For obvious differences, move quickly between very low and very high Ratios. Ignore the metering and listen to how it influences the transients, reducing the Threshold to emphasise the effect if necessary. Focus on the subtle differences between lower Ratios such as 1.5:1 and 4:1.

**5** > Because compression works by reducing the level of signals, it is often necessary to raise the output level or Makeup Gain to compensate for the loss in volume (this does not change the compression's behaviour or character). Repeatedly bypass the plugin, adjusting Makeup Gain with the goal of getting the compressed signal to a similar level as the bypassed one.

**6** > Some compressors offer enormous flexibility, allowing super-fast attack/release, various detector modes, and sidechain filtering. Live's Compressor is a case in point, and here we're using its very fastest settings combined with automatic Makeup Gain. Note how easy it is to introduce clicks and distortion, so approach these processors with care, unless, of course, this is what you want.

## The classics

**A handful of classic hardware compressors have won through over the years, and you can find the vast majority of these in software form. If you're new to them, though, it can be tough to know just how to dial in 'that' sound, so let's take a look at three bona fide compression legends and give you some pointers.**

**The Urei 1176 is an FET (Field Effect Transistor) design heard on countless hit records. It has simple controls (input, output, attack, release and ratio) but can achieve extremely fast attack and release times. With fast settings, it's great for adding bite to vocals and bass. With a slower attack, fast release and high ratios, it creates amazing pumping drum overheads. Try emulations from Universal Audio, IK Multimedia, Native Instruments, Softube and more.**

**SSL's Bus Compressor is best known for its glue-like effect when applied to the entire mix via the master buss. Try the 4:1 ratio with a slow attack (30 ms) and a fast release (0.1 sec). Alternatively, use the 2:1 ratio (which has a softer knee) with a slow attack (10 or 30 ms) and the auto release. In both cases, aim for gain reduction of between 2 and 4 dB. Emulations can be had from SSL themselves, Waves, Universal Audio, and many more.**

**The valve-based Fairchild 670 commands big bucks second hand. It can work in stereo and mid/side mode and has unusual parameter labeling, such as six 'time constant' settings that handle attack and release (1 is fastest, 4 slowest and 5 and 6 more program dependent), and a DC Bias screw for ratio (2:1 to 30:1). In use, it can be great on drum sub groups, vocals and, if you can just tickle the gain reduction, on the mix buss. For emulations, check out Universal Audio, IK Multimedia, Slate Digital and Waves.**

# The right compressor for the job

As touched upon on in our mini-roundup of classic units, not all compressors are born equal. But why do they sound different at all? And how do you know when to use which 'type'?

First up, going back to hardware devices, there are a few different types of detector/gain control circuit: **VCA** (Voltage-Controlled Amp), **FET** (Field Effect Transistor), **variable mu** (based on valve plate voltages) and **opto** (lamp plus photocell). VCA designs can be very flexible but often the flexibility makes them harder to use. FET designs are responsive but distort sooner, although that distortion is very musical. Opto designs have a softer, more musical attack and tend to let fast transients slip through. Finally, variable mu can be very fast in attack, however threshold, ratio and knee shape tend to all be very closely interrelated. Some units also have an **automatic release** option, where the release time changes to suit the incoming signal.

By now, you're probably a little bewildered about where to start when compressing your own signals, but worry not, as we've provided suggested 'starting point' settings to help you pick the right compressor for the job and get a basic sound on the go.

## Compression settings

**ATTACK:**
**Super fast** - below 1ms
**Fast** - 1ms to 20ms
**Medium** - 20ms to 75ms
**Slow** - 75ms or longer

**RELEASE:**
**Fast** - fastest to 20ms
**Medium** - 20ms to 200ms
**Slow** - 200ms or longer

### Voice

VCA ● FET ● OPTO ● VARI-MU

| ATTACK | RELEASE | RATIO | GAIN MAKE-UP | GAIN REDUCTION |
|---|---|---|---|---|
| FAST | MEDIUM | 4:1 | AUTO | 4-6dB |

### Snare drum

VCA ● FET ○ OPTO ○ VARI-MU

| ATTACK | RELEASE | RATIO | GAIN MAKE-UP | GAIN REDUCTION |
|---|---|---|---|---|
| FAST | SUPER FAST | 4:1 | AUTO | 4-10dB |

### Acoustic guitar

VCA ● FET ○ OPTO ● VARI-MU

| ATTACK | RELEASE | RATIO | GAIN MAKE-UP | GAIN REDUCTION |
|---|---|---|---|---|
| FAST | MEDIUM | 4:1 | AUTO | 2-6dB |

### Drum kit overheads

VCA ● FET ● OPTO ○ VARI-MU

| ATTACK | RELEASE | RATIO | GAIN MAKE-UP | GAIN REDUCTION |
|---|---|---|---|---|
| MEDIUM /SLOW | FAST | 8:1 | AUTO | 6-8dB |

### Bass guitar

VCA ● FET ○ OPTO ○ VARI-MU

| ATTACK | RELEASE | RATIO | GAIN MAKE-UP | GAIN REDUCTION |
|---|---|---|---|---|
| MEDIUM FAST | MEDIUM FAST | 8:1 | AUTO | 3-6dB |

### Kick drum

VCA ● FET ○ OPTO ○ VARI-MU

| ATTACK | RELEASE | RATIO | GAIN MAKE-UP | GAIN REDUCTION |
|---|---|---|---|---|
| MEDIUM | MEDIUM /FAST | 4:1 | AUTO | 4-6dB |

### Picked electric guitar

VCA ● FET ○ OPTO ○ VARI-MU

| ATTACK | RELEASE | RATIO | GAIN MAKE-UP | GAIN REDUCTION |
|---|---|---|---|---|
| FAST/ SUPER FAST | FAST | 8:1 | AUTO | 6-8dB |

### Master bus (edgy)

VCA ● FET ○ OPTO ○ VARI-MU

| ATTACK | RELEASE | RATIO | GAIN MAKE-UP | GAIN REDUCTION |
|---|---|---|---|---|
| MEDIUM | FAST | 4:1 | AUTO | 4-6dB |

### Lead synths

VCA ● FET ○ OPTO ○ VARI-MU

| ATTACK | RELEASE | RATIO | GAIN MAKE-UP | GAIN REDUCTION |
|---|---|---|---|---|
| MEDIUM /FAST | FAST | 4:1 | AUTO | 4dB |

### Master bus (smooth)

VCA ● FET ○ OPTO ● VARI-MU

| ATTACK | RELEASE | RATIO | GAIN MAKE-UP | GAIN REDUCTION |
|---|---|---|---|---|
| MEDIUM | AUTO | 2:1 | AUTO | 2-4dB |

# Frequency-conscious dynamics

As touched upon previously, some dynamics processors have filtering or EQ in their sidechain, causing them to react more (or less) to certain frequencies. This is known as frequency-conscious dynamics processing, and the most common form is the de-esser, a tool used to control high-frequency sibilance in vocals. However, the same concept is also found in general purpose frequency-conscious dynamics and dynamic EQs.

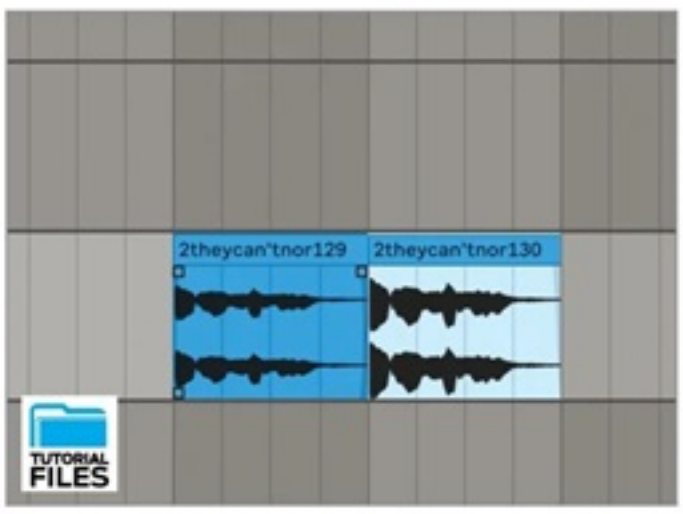
For a de-esser or frequency-conscious compressor, the filtered sidechain signal feeds (aka keys) the detector, so gain reduction is caused predominantly by the focus frequencies. In the case of a dynamic EQ, the sidechain keys the gain of a filter circuit, and so it's basically a specialised EQ and we therefore won't be covering it in this article as we want to focus on the core dynamics processors (note, however, that some de-essers do use this technique of dynamically controlling EQ filters).

In a basic de-esser, 'ess' sounds will trigger broadband (full frequency) compression, which will cause the whole signal to be reduced in level during the sibilant parts. However, by using a filter circuit to split the signal, it is possible to process just the relevant frequency range, leaving the rest unscathed, and some de-essers offer this functionality – in essence, what you're seeing is a basic multiband dynamics processor (which we'll cover in detail on page 48).

> "In essence, what you're seeing is a basic multiband dynamics processor"

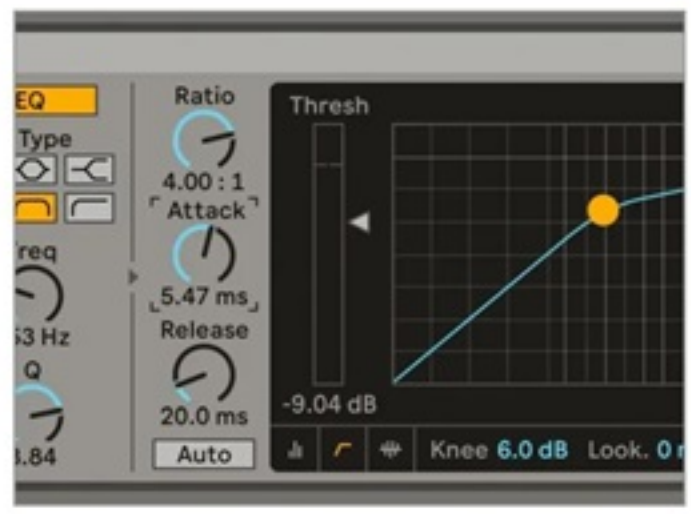
A more general-purpose frequency-conscious compressor simply takes the idea of the dynamic de-esser and provides a fully flexible frequency range and filter shape for the key signal, and often you'll find this concept incorporated in an existing compressor (Waves' C1 for example). At a slightly simpler level, you often get compressors with an adjustable high-pass option on the sidechain, and this can help reduce sensitivity to low-frequency content, such as when you want to compress a drum loop without the kick drum causing the compressor to over-react.

## > Step by step 2. Frequency-conscious dynamics processing

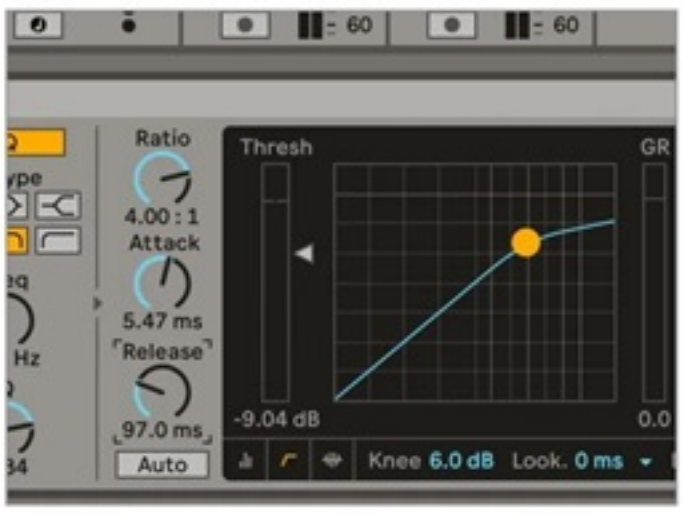



1 > Here we have a vocal in which the low frequencies jump out too much at various points – a problem that we can fix! Frequency-conscious compression is the tool of choice here. Load up a compressor with flexible sidechain options – we're using the excellent stock Compressor in Ableton Live, with the extra options panel opened via the disclosure triangle.

2 > We turn on the sidechain EQ and select the band-pass shape, allowing only a narrow range of frequencies through. The sidechain listen option (headphones icon) lets us hear the filtered sidechain signal in isolation. Sweeping the Freq knob, we find the boominess at around 150Hz. We narrow the Q to emphasise the problem area in the sidechain signal.

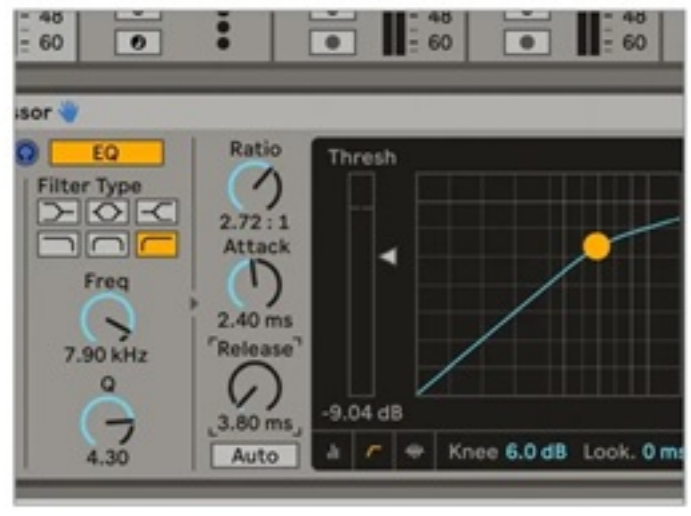



3 > We switch out of sidechain listen mode. The compressor now reacts more strongly to those lower-mid frequencies, but we still need to address the actual compression settings. We set the Ratio to 4:1 and lower the Threshold until we get some compression. Then we adjust the Attack and Release – a fast Attack makes for a good starting point.




4 > Dial in a medium Release setting. Remember, although the compressor is responding mainly to the lower frequencies, it is still acting on the full frequency range. If you get too aggressive with your compression envelope settings, the signal will pump too hard, overshadowing your efforts to tailor the response. For low frequency problems, you can expect to achieve a few dB of correction with this approach.

5 > We can use the same technique to create a de-esser. Load up a second compressor, again set to the band-pass filter shape, and this time use the sidechain listen to focus on the high frequencies from 9-11kHz. Once you've found the sibilance, adjust the Threshold, Attack and Release to shape the compression to taste.

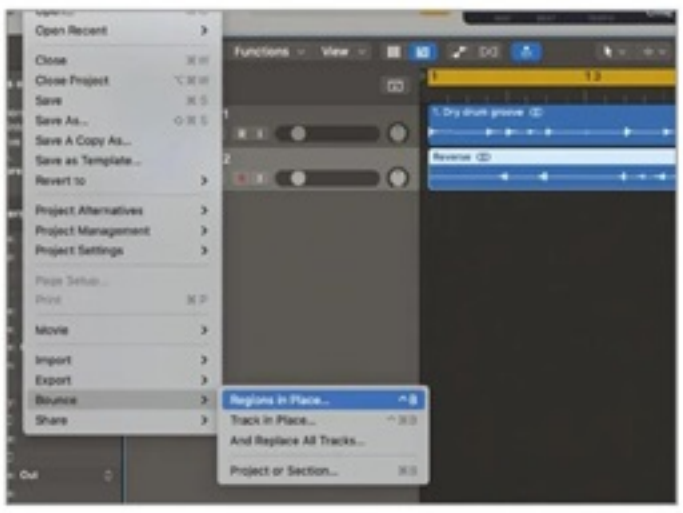



6 > Sometimes, a band-pass sidechain filter can be tricky to fine tune for de-essing. A simpler alternative is to use a high-pass Filter Type instead, which gives a broader response. Change the sidechain filter type and switch to sidechain listen mode to focus the cutoff point (start at around 8kHz) until you hear mostly sibilance. Then switch out of sidechain listen mode and adjust the Threshold, Attack and Release as before.

## > Step by step 3. Creative processing with reverse compression

**1** > Let's now look at compression in a creative context – for an unorthodox flavour, try reverse compression. This is similar in execution to reverse reverb, and pretty easy to achieve within any DAW. The first step is to copy the audio file to be processed and reverse it, then put it on a 'temporary' track with a compressor inserted into it.

**2** > Here we're processing a drum kit loop. Remember, since our audio is in reverse, the Attack setting will ultimately influence the decay stage, while the Release will become the attack stage. With the Release set fast, then, we can retain the transients; and by using reasonably heavy compression, we can add some colour to the release stage.



**3** > With the Attack and Release set, simply adjust the Threshold to achieve the desired amount of gain reduction – we've gone for 10dB. Then bounce the processed audio to a new file, place it on the original track and reverse it so that it plays the right way round.

**4** > While this process is pretty quick, as with reverse reverb, the quality of results is largely down to trial and error, and very dependent on the source audio, so it might take a few goes before you get something you're happy with. Here, as an alternative, we're using more laid-back Attack and Release settings.

## External sidechaining

**So far we've talked about the built-in sidechain that takes the compressor's input signal, splits it in two, then sends one half (possibly with additional filtering applied) to the detector and the other half to the gain adjustment section. It is, however, possible to use entirely separate signals for this by using what's known as an external sidechain, where a different signal is routed to key the detector and decide when the level is affected. For instance, place a compressor on your bass channel, and send the kick drum to its external sidechain input, and whenever the kick drum is played, compression will be applied to the bass sound. This has the effect of making the bass drop in volume to 'duck' out of the way of each kick drum hit.**

**As well as being useful from a mixing point of view, to help sounds sit neatly together, the effect can also be used overtly, for an obvious effect. Indeed, perhaps the most important sound in dance music over the past decade has been that of sidechained compression. To get the best results, you'll need to fine-tune the attack, release and threshold to get the desired pumping quality.**

**The same technique can be applied in other ways. On a reverb return, for example, set up a compressor sidechained from the kick drum for a pumping reverb you can add to all sorts of sounds. If the reverb is used on your kick/snare, try sidechaining it from the same sound you're sending to the reverb. This will carve out a little dip for the attack.**

## > Step by step 4. Phattening drums with snap compression

**1** > Compressing drums is a balancing act: you're aiming to control the transients without crushing them. 'Snap' compression is a great technique for retaining the transients but still getting a fatter overall sound. It's particularly good for snare drums, rimshots and – sometimes – kick drums. Start by soloing a snare and slapping a compressor on it.

**2** > We need to set the Attack time just slow enough that the transient can make it through the compression unscathed. Doing this by ear, we end up at around 50ms. We then select a 6:1 Ratio and lower the Threshold until we get moderate gain reduction of around 8dB.

**3** > The idea is to compress the snare tail immediately after the transient, to make it feel punchier, and 'flatten' the tail, giving the impression of increased sustain. So, the final job is to set the Release very fast and readjust the Threshold to taste. The result is a much fatter snare drum with enhanced transient 'crack'.

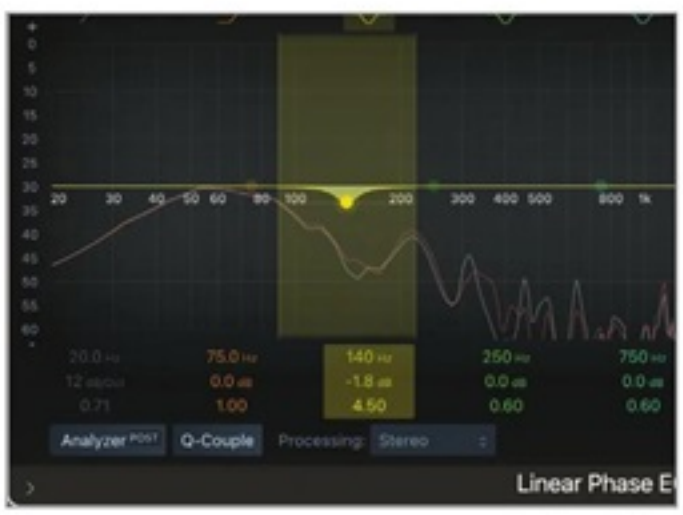
## > Step by step 5. Retaining dynamics with parallel compression

1 > Parallel compression involves mixing heavily compressed and uncompressed versions of the same signal. Here we're taking the traditional approach, using two separate channel, but you can also do it using a compressor with a dry/wet mix control. First, copy the part to be compressed over to a second track (another option is to bus it to two channels in your mixer).

2 > Decide which will be the compressed track and which will be your dry track, then load your compressor of choice onto the former. We want to push the compression quite hard, so set a fast Attack, medium Release and high Ratio, then lower the Threshold until you get an obviously squashed, fat sound.

3 > Now blend in the unprocessed track. Starting with the fader down low, gradually pull it up until you start to hear the sounds combine. Make a mental note of this starting point, then work up from there to try different balances.

4 > Our plan is to combine the colour of the heavy compression with the openness of the unprocessed source. There are no hard and fast rules, but once you have a balance you're happy with, try muting and unmuting the unprocessed track to check that it's working as envisioned. Be prepared to adjust the settings on the compressed track, too.




5 > Parallel compression can be further enhanced with some careful EQ. To avoid phasing problems when mixing EQed and non-EQed signals (or indeed, two EQed signals), we recommend using a linear phase EQ. Here, we're using a low-mid EQ on the unprocessed channel to modify the body of the snare, typically found around 100-400Hz. Here, we've set up a bell-shaped cut of about -3dB.

6 > Our final move is to route both tracks to a group bus, in order to process them together before they reach the master output. This time we use a regular, non-linear phase EQ, sweetening the sound with gentle low and high shelves.

## Parallel compression

**Parallel or 'New York' compression is simply the concept of splitting a signal at source into two parallel paths, compressing one while leaving the other unscathed. Once you blend the two back together, you should, in theory, be combining the benefits of compression while retaining the air and cleanliness of the source.**

**However, in truth, the technique tends to be used to take advantage of the colour a compressor can add, so in many cases you'll be pushing the compressor harder and then blending the unprocessed signal back in with that. You'll typically see it used for processing transient heavy sounds such as kicks and snares, and also for fattening up lacklustre vocals. Get it right and the results will differ from simply using the compressor on its own. But it can be a fiddly technique, and it's very dependent on the level balance between the two signals.**

**There are two ways to approach it: one is the old-school method, originally developed on analogue desks where you would split (or 'mult' to use the US terminology) the source down two desk channels. On one channel you can then insert your desired compressor, setting it to compress hard, bringing up the tail of sounds and generally fattening them up. On the second channel you can then blend back in the raw source, regaining some of the lost transients or in the case of a vocal, articulations. So far so good. But the technique can then be further finessed by EQing each channel slightly differently. In a DAW environment, one can use linear phase EQs to minimise phase smear between the two channels.**

**The second way to approach parallel compression is to use a compressor with a mix blend (dry/wet) control. This is much quicker and can also give you an idea of whether the technique is worth a try. However, you clearly miss out on the additional EQing option, and achieving an accurate balance using a 'mix blend' is arguably much harder than blending two fader levels.**

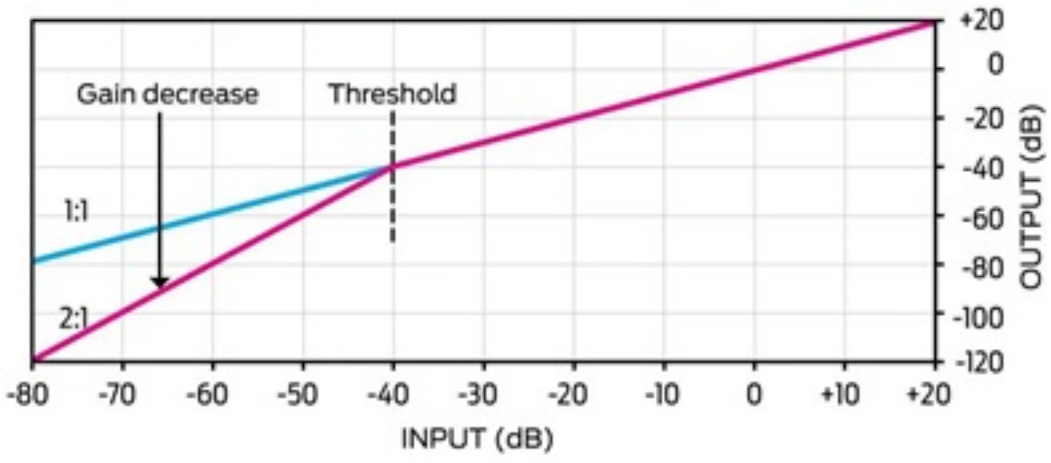
# Expansion

An expander increases dynamic range, and like a compressor can work in both downwards and upward ways. An upward expander increases gain above a threshold, ie, the opposite of a standard compressor. However, in practice, we typically encounter them as downwards expanders, and in this format, any signal falling below the threshold is reduced in level.

The amount of gain reduction is again based on the ratio, and this is typically very gentle – 2:1 or 3:1, maybe. You'll find other typical dynamics controls such as attack and release, but also some that are less common elsewhere such as **hold**, which sets a minimum period to sustain the gain reduction before the release can begin, and a **range** or **limit** (not to be confused with limiters and limiting) control, which affects the overall maximum gain attenuation applied – useful when trying to make the expander sound as natural as possible.

If we plot a downwards expander on a transfer graph, we can see the graph become steeper at the threshold, and the slope of the steeper plot will be dependent on the ratio. At a ratio of 2:1, if the signal falls 3dB below the threshold, the output signal will be reduced by 6dB. In practice, the levels will be changing over time along with gain reduction.

So, what can we use them for? Typically, the curtailment of single-ended noise like tape hiss, mic bleed or background noise. We say curtailment, as they won't remove these sounds completely; ideally, they'll do it in a more subtle way. Next, we'll show you how to fine-tune an expander's parameters to get the desired result.

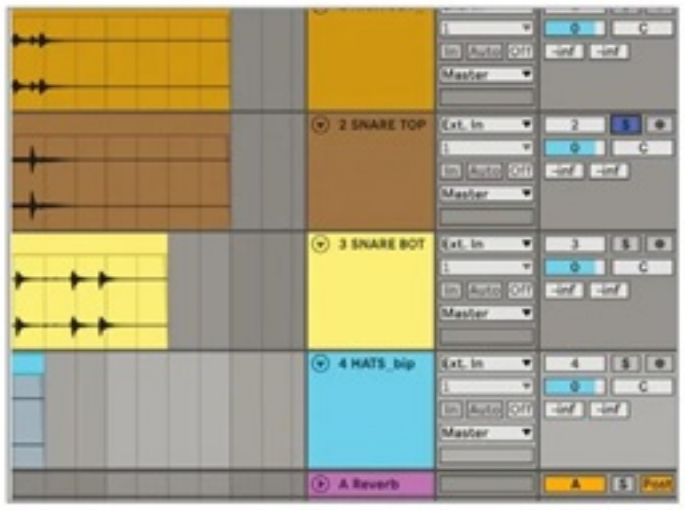



**A downwards expander will reduce the level of a signal below the threshold**

## > Step by step

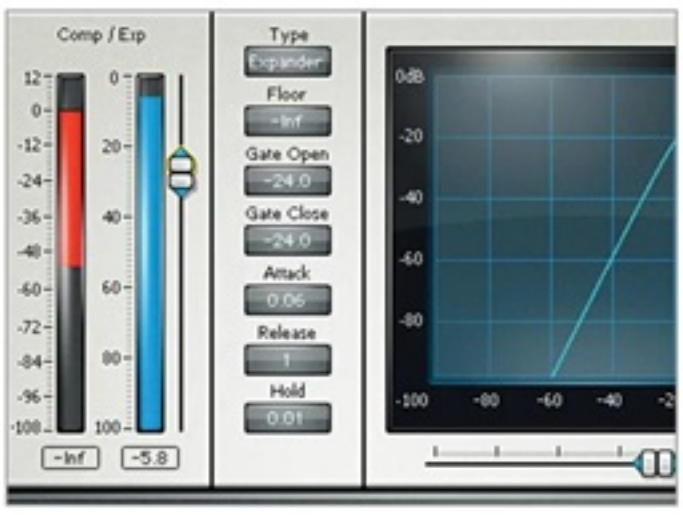
## 6. Reducing background noise with expansion

**1** > Expanders are great for suppressing background noise or ambience in situations where a gate would sound too abrupt – drums or multi-miked location recordings, for example. Here we're going to use them to curtail – but not entirely remove – mic bleed from a snare drum's underside mic. Load the drum tracks in Tutorial Files into your DAW.




**2** > Solo each track to hear how the bleed between drum mics potentially makes controlling the sound quite tricky – when we start to introduce EQ and compression, any bleed will likely get magnified, which could seriously limit our options as well as clogging up the mix. If we gate them, though, the sound will be too clinical, so an expander is our best solution.

**3** > Let's expand the bottom snare mic, which is picking up bleed from the kick as well as the snare wires rattling in sympathy with the other drums. Load an expander and set a fast Attack and Release. Our expander here has a fixed Ratio of 2:1 and a Floor, or gain reduction limit, which we'll come back to shortly. Set the Floor limit to maximum for now – on our plugin (Waves C1), that's -infinity dB.

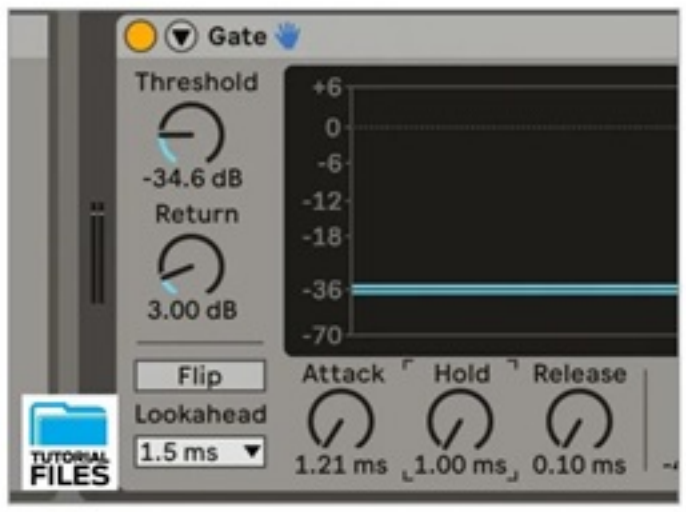


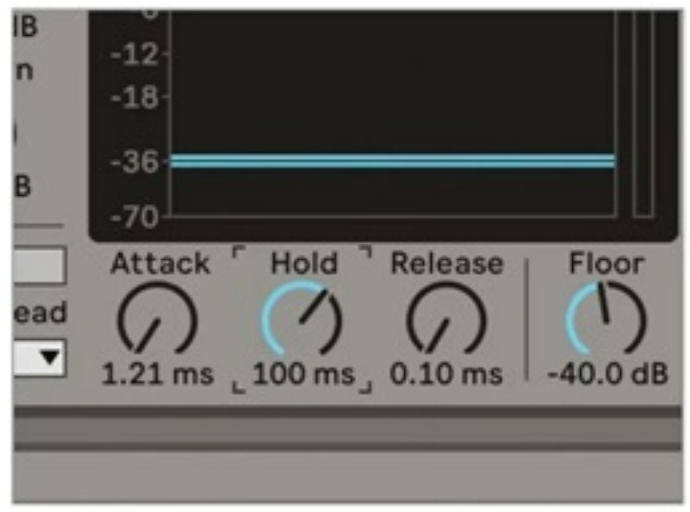

**4** > Gradually raise the Threshold from its lowest setting to 0. You'll notice that the gain reduction happens in the gaps between the louder sounds, which is exactly the effect we're trying to achieve. Keep moving up until it sounds choppy and gated, removing most of the sound between the hits.

**5** > We started with a fast Release setting, but now let's ease it back to make the sound less choppy and gate-like, and allow the sound to breathe a bit. We've gone for around 200ms, but the actual setting will depend on the source signal and the style of music, so let your ears be the judge.

**6** > Finally, back to that Floor limit parameter. The current setting lets the expander reduce the signal right down to -infinity dB – ie, silence. We can adjust this to set the maximum amount of gain reduction applied, so that the expander quietens the gaps between hits rather than totally silencing them. We go for a 20dB limit. The last thing to do is tweak the Threshold once again to finesse the sound, then bypass to compare.

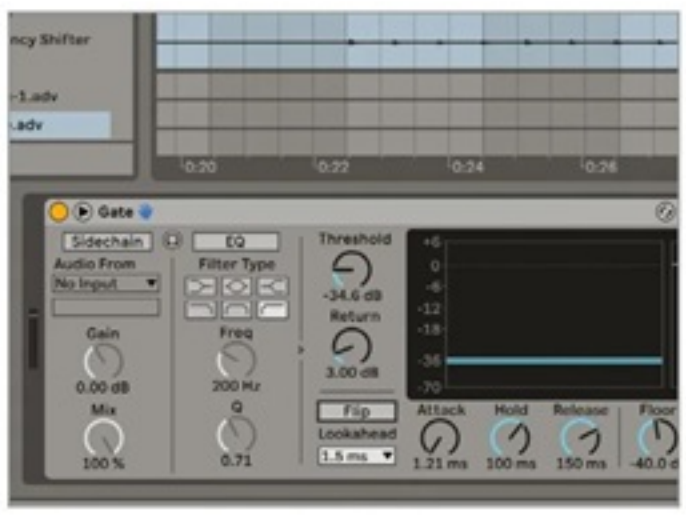
## > Step by step 7. Basic noise gate setup




**1** > Gates differ from expanders in that they work by switching between two states – open and closed – rather than responding continuously to the signal level. Here, we're gating a snare drum to remove mic bleed and ghost notes. First, set a fast Attack, then adjust the Threshold upwards from minimum until the gate only opens for the main hits.

**2** > Now adjust the Hold time, which sets how long the gate stays open once the level drops back below the Threshold. It's great for drums, enabling accurate timing and preventing rapid opening/closing caused by levels hovering around the Threshold. Here, around 100ms does the trick. Be prepared to adjust the Threshold as you tweak the Hold time.



**3** > Now adjust the Release setting to taste. Here, around 150ms sounds natural, but 30ms delivers a good, tight sound. You may also have a Hysteresis setting, which is an additional closing threshold. Adjusting this below the opening Threshold helps if the gate is 'chattering' (opening and closing too much), but as discussed, you can also resolve this with the Hold setting.

**4** > Finally, Reduction (or Range or Floor as shown) sets the attenuation when the gate is closed. Completely silencing the signal can sound unnatural, so reducing this is like blending in some of the 'dry' signal. For more accurate results, use a gate with sidechain filtering, which can be configured in much the same way as frequency-conscious compression.

## Noise gates

**Like most of our dynamics processors, noise gates started out as functional devices. As noted, they differ from expanders, as when the signal is below the threshold, the output level is reduced by a fixed amount – and possibly to silence – rather than by a ratio. However, be aware that a gate can be a blunt tool, as its operation can sound very obvious, particularly if it opens in the wrong place or doesn't open when required.**

**Functional uses include quick tidying up of recorded tracks to remove unwanted performer noises in between playing, removal of tape or ambient noise in between the desired audio, and automatic muting of mic channels when recording or in a live setup. Some gates include a reverse 'ducking' mode, so they close rather than open during the presence of a signal.**

**Gates are also great for creative applications. With a very fast gate, you can use the attack setting to add a click to the front of a sound and use the release to shorten its decay. They're also great for gated reverb, using the hold and release to tailor the reverb tail so it drops suddenly after an initial decay phase. Finally, using their sidechain, gates can be used to impart the rhythm of one sound on another. This can be great for locking two live instruments together or for applying specific patterns to a sustained sound in a 'trance gate' fashion.**

## > Step by step 8. The classic gated reverb effect

**1** > Gated reverb is a classic effect that's really easy to set up. Start by loading a reverb onto an auxiliary send channel. After we've applied the gate, we'll only be hearing the early part of the reverb tail, so use a reverb that has a big sound in its early stages and – unless you want a slap-back effect – zero the Predelay.

**2** > Load a gate onto the reverb return channel, placed right after the reverb. Dial in fast Attack and Release settings, and set the Hold to around 100ms. Adjust the Threshold so that the gate opens as required, then return to the Hold and Release settings and tweak to get a sound you like.

**3** > Like delay, gated reverb really needs to be timed accurately. You can either do this by ear, or work the timing out mathematically. With our track at 105bpm, a quarter note is 286ms in length, so we can set our Hold time to around 286ms to start with, or use 286ms as a target for the Hold and Release times in tandem.

# Transient processors

Transient shaping was made popular by SPL with their classic Transient Designer hardware unit, which uses something called 'differential envelope technology' to shape the transients of sounds no matter whether they're quiet or loud ones. In fact, this is the only processor we're covering that doesn't react to level directly – there's no 'threshold' – responding instead to *changes* in level.

SPL's processor is now available in plugin form both direct from SPL and also via the UAD platform. It's no longer the only such plugin on the block, though, as you'll find transient shapers from a number of plugin manufacturers. They tend to fall into two camps: the simple two-knob attack/sustain types, and the graphic-based 'envelope shapers' which tend to include a few more parameters. Either way, what we're looking at is a processor that tracks and reshapes the envelope of the signal it encounters.

It's tempting to just head for the attack control to adjust the front end of a sound, but you also have control over the sustain, and this can be just as important when finessing the sound. Reducing the sustain for a more 'clipped' (as in abrupt, not distorted) sound can be a great alternative to gating when tightening up beats. However, increasing the sustain is often more rewarding, and on drums, for example, you can create the impression of quite heavy pumping compression simply by extending out the sustain control, without the transient-squashing that can be associated with compressors. In general, transient shapers work best on shorter sounds, as smooth, sustained sounds can begin to sound a bit lumpy under heavy processing. Finally, watch your output levels, as a transient shaper can generate considerable peak levels.

> "Reducing the sustain for a more 'clipped' sound can be a great alternative to gating"

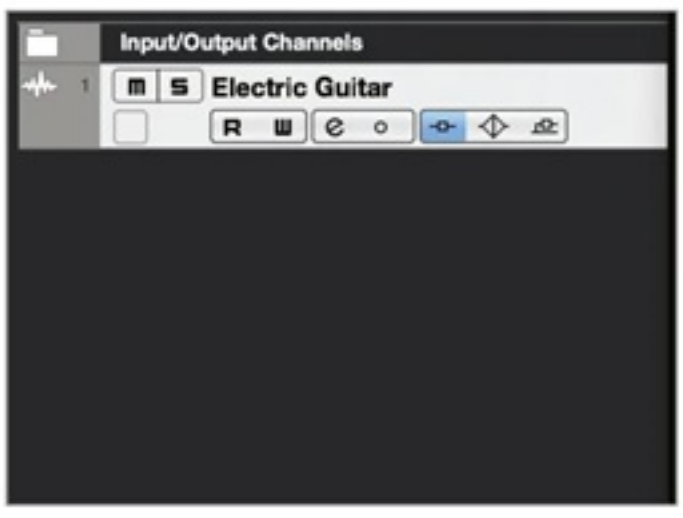
In the walkthrough below, we'll try two things: first, we'll enhance the attack of a snare drum, and then we'll soften off the attack of a picked electric guitar. In both cases, once the attack is set up, we'll find that the sustain also plays a vital part in fine-tuning the outcome.

## > Step by step 9. Exploring transient shaping processors

1 > Transient shaping plugins are related to compressors and expanders but work somewhat differently. Let's start with something simple: adding sharpness to the attack of a snare drum. Our plugin (SPL Transient Designer) allows +/-15dB of cut/boost, but it only takes a 3-4dB Attack boost to get a crisper sound.

2 > What happens if we push it further? Over about 10dB, the snare really starts to pop. Not only that, but the plugin's 'clip' lamp (labelled OVR on this plugin) comes on, indicating that the signal has exceeded 0dB, and the sound quality starts to suffer. We trim back the Output Gain level to compensate.

3 > Next, let's investigate what the Sustain control can do for our snare. With our big Attack boost still dialled in, pushing up the Sustain level actually brings some life back into the sound. However, pulling back the Attack slightly sounds even better. The overall effect is similar to 'snap' compression, but easier to achieve and without as much level adjustment.




4 > Cubase's EnvelopeShaper transient processor features just three main controls – Attack, Length and Release – although you can also drag the envelope points in the display. Let's use it to soften the attack of a clean-picked electric guitar. We start with the default preset.

5 > Our first step is to reduce the Attack parameter. A little goes a long way here, and it's quite easy to end up with what sounds like a fade-in on each note. We've opted for a very small adjustment, but bypassing and A/B comparing reveals that the result is exactly what we want.

6 > We can tweak this further by dragging out the Length and reducing the Release time slightly. Once again, we've reshaped our sound in a very direct way, and although we could have used compression to soften the attack and influence the release of our guitar, that's a much blunter tool for this sort of process.

# 10 quick tips: EQ + dynamics

There are many ways to lift and polish mixes with EQing and dynamic processing

01 EQ sweeping is a technique that helps locate problem frequencies and find sweet spots. Select a bell-shaped EQ with a fairly sharp Q, then add a boost of 12dB or more. Sweep the frequency control until the element you want to adjust 'pops out'. For musical parts with changing notes, you may have to do this a few times to home in on them. With frequencies established, you can confidently apply more musical boosts or cuts.

02 When EQing bass, a useful trick is to combine a high-pass filter with a low shelf or bell. This gives you the flexibility and musicality of the low band with the tailoring precision of the filter, and helps keep the very low frequencies in check. A good starting point is to set your filter at around 50Hz with a medium slope. Keeping an eye on your frequency analyser or monitoring with a subwoofer can really assist with decisions here.

03 To create rhythmic sounds from any audio, the easiest way is to use a sidechained gate. The gate itself will include a threshold, attack, release and hold time, allowing you to specify how it opens and for how long. Meanwhile, a rhythmic key triggers it to open. By using an instrument as the key you can either program or (for more fun) actually play the keying signal, making the whole process pretty creative too.

04 There are times when you may need to curtail frequencies dynamically, so that they're only controlled when they happen. The most typical example of this is a de-esser. However, more general frequency-conscious processors can be used anywhere. Dynamic EQs should not be overlooked – these will typically allow you to solo and specify the key signal frequency and then set the compression amount all within one interface. Often the compression circuit will allow you to select how broad the action is, offering a good deal of precision.

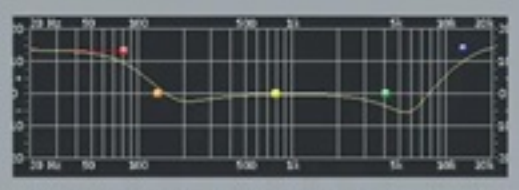
**Sweep a band with a sharp Q or use a frequency analyser to identify problem frequencies to be reduced**

05 Powerful parametric EQs are great for sorting out honking frequencies or loud notes. Use EQ sweeping or a frequency analyser to fine-tune the point (or points) and then cut with a sharp Q. Deep, sharp cuts can work well without affecting the surrounding frequencies.

06 Compression adds flavour but can reduce transient definition and punch. For the best of both worlds, try 'parallel compression': split the signal into two channels and compress only one, then blend them together to balance flavour and punch. Some plugins have mix controls for this, but using twin channels is more flexible.

07 Although it's possible to use automation to 'ride' levels of competing mix elements, you can also insert a gate on the track that you want controlled, and then key it from the sound that you want to keep audible. It's great for ducking sounds such as guitars or lead synths when they're competing with vocals.

**Use a dynamic EQ such as this one by Brainworx to reduce problem signals at the right time only**



**Take advantage of EQ overshoot to create more spacious boost/cut shelves**

08 Shelving EQs vary in shape with many lifting frequencies very broadly. Sometimes you'll want this, but it can also create too much lift. This is where an 'overshoot' shape can come in handy, as increasing the Q setting will eventually lead to a dip just before the lift, creating vital frequency space.

09 Reverbs can occupy lots of frequencies, and in isolation this can sound very impressive. But in reality, it can make things muddy too. By using a low-pass and high-pass filter after your reverb return, you can quickly tailor the sound without getting bogged down with the plugin's own parameters.

10 A compressor can be a great tool for shaping the envelope of a sound, especially drum beats. By using a slower attack and faster release alongside an auto gain make up option, it's possible to retain some attack, and almost 'clamp' the remainder of the sound, for a punchy effect known as snap compression.

# Sidechain compression

> What started out as a rare mix technique designed to add glue between specific sounds has become, without question, the most popular audible compression technique ever, and much like extreme pitch correction, is often overused. That said, things have clearly moved on from the early days, and applied with care, the process can add life and energy to your tracks without killing the dynamics.

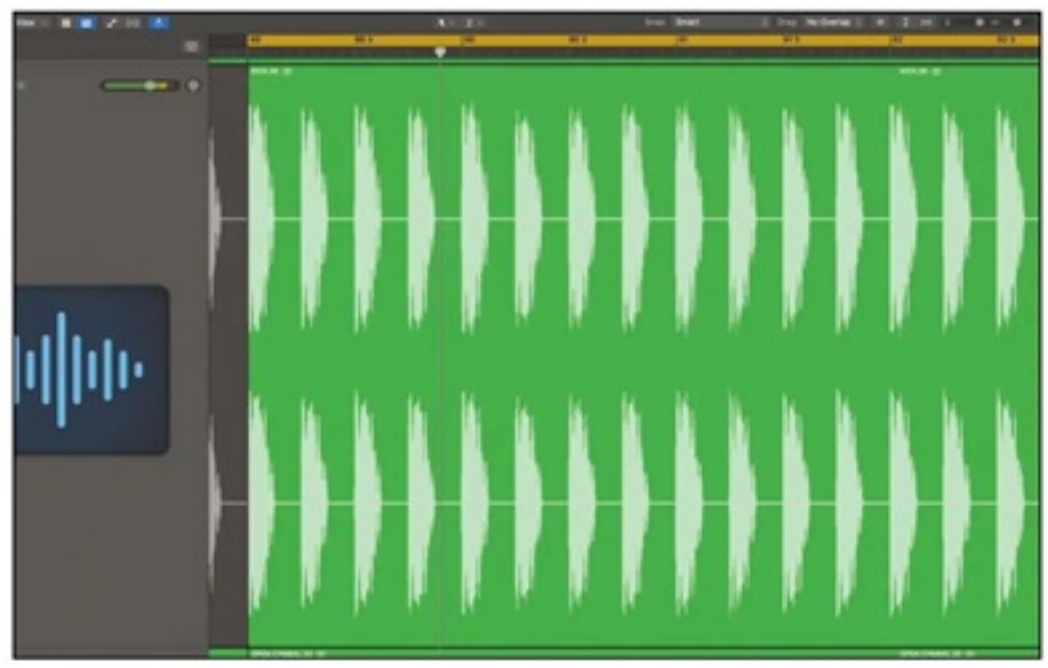
But what exactly do we mean by the term sidechain compression? In fact, all compression actually uses a sidechain, it's just that it normally comes from a split of the compressor input signal itself, and this is why the compressor reacts to the input. However, if we can use an external signal of our choice to instigate the compression, then it opens up a whole world of creative opportunities as the compressor will apply the dynamics of the external signal to its own input.

Of course DAW flexibility allows us to be quite specific with how we set this up and allows us to not only process specific tracks or subgroups, but also create bespoke sidechain blends. Throw in some automation as well and we have a highly flexible way to shape dynamics.

Here we'll show you how to set up a typical pumping sidechain compression effect, suggesting routings and basic settings. Then we'll look at how to finesse it and make it do what you want. Finally, although the focus of this tutorial is compression, if you're simply after a rhythmic pumping effect, this can be created in other ways, and there are even some bespoke plugins (CableGuys Kickstart 2 for example) that do this. With this in mind, we'll finish up by looking at one specific and quite simple technique that can create sidechain compression pumping effects.

## > Step by step 10. Sidechain compression

1 > To get started with sidechain compression, you need a compressor plugin which includes an external sidechain input. Insert this plugin across whatever you want to process. This could be individual tracks, submixes or the whole mix. Here we're using it on just the pad sound from our mix.



2 > Next up, you need a sidechain or 'key' signal. You can use anything for this, and even create a bespoke track just for the key signal. For a typical rhythmic pumping effect you'll need something with a clear transient that is also in time with the track. For club music we usually use the kick.

3 > Next up, you need to check how your particular DAW routes signals to the compressor sidechain. The simplest option, something which is available on many DAWs, is to select a specific track as the source. Here we've selected our kick track from the compressor side chain menu.

4 > It's worth saying that as an alternative you can create a separate bus for the sidechain. Send whichever components you like into this bus and select the bus as the sidechain signal on the compressor. This is quite a flexible option and a great technique if you want to control the compressor from multiple sources.

## > Step by step 10. Sidechain compression *(continued)*

5 > Back with our simple kick sidechain, we are now going to set the compressor to shape the pad sound. With a default preset loaded, set the ratio to about 8:1 and lower the threshold until you start to get some healthy gain reduction – we're seeing about 15 to 20dB. This should create audible rhythmic pumping.

6 > Now finesse the effect. If you have the option, set the knee shape to hard. Then look at the attack and release times. Set the attack to moderately fast (ours is 15ms) then focus on the release. With the fastest release times you get the most pumping, so use this to adjust how dramatic the effect is.

7 > Once you have an effect you like, you may want to adjust its overall depth. Rather than mess with your carefully crafted attack and release times, try adjusting the ratio. Alternatively, many compressors also have an output mix, offering a second way to achieve your desired effect blend.

8 > As mentioned, you can get a pumping effect that sounds like sidechain compression using other techniques, including bespoke plugins. If you have a flexible stereo level modulation effect that can be set to tempo sync this can also work. Here we've loaded up the Tremolo effect in Logic Pro X.

9 > By default this plugin uses alternating left and right level changes to create the tremolo effect, but simply by setting the Phase offset to zero, the plugin now applies the level change evenly on both sides. With the Rate set to a quarter note, we have something approaching a pumping effect.

10 > If we now set the curve Offset to 180 degrees and Smoothing to 0%, the initial level is zero. Now by adjusting the overall Depth (we've set ours to 35%) and introducing a small amount of Smoothing (we've set ours to 10%), you have a time synced pumping effect, which emulates sidechain compression with a fixed quarter note trigger.

# Serial compression

Like many dynamics processes, compression can be a tricky one to get right, particularly where levels are quite variable. Too little compression can leave you wondering why you've bothered, too much and you're subsequently saddled with unpleasant artefacts or unwanted pumping.

Couple this with the desire to make use of a compressor's inherent flavour and you might begin to wonder how you'll ever get where you want to be. One very effective technique is to stop trying to achieve multiple outcomes from one processor and instead use two or sometimes more compressors in series.

Serial compression allows us to play to the strengths of each processor. We could, for example, try combining the glue effect of an opto compressor with the transient modification and distortion of a FET. Or maybe experiment with the harmonic distortion of a valve design and the predictability of a VCA. Or we could simply use one compressor to smooth out levels in combination with another that can grab peaks. In most situations we are also spreading the overall gain reduction, and this is ideal in more critical scenarios such as mastering, where we may not want the audio to actually sound processed at all.

Of course using multiple compressors then requires us to decide which order to put them in. Usually the required task will help us decide. That said, creative techniques like this are always open to interpretation, and it's often worth trying different orders. Finally, serial compression isn't a technique that's suited to every application. What's more, although it can be a good way to handle tricky levels, we wouldn't advise its use when recording, as setting multiple thresholds can be fiddly. Ultimately, serial compression can be a highly fruitful technique at the mixing stage when you have plenty of time to properly finesse settings.

## > Step by step 11. Serial compression

1 > Serial compression offers a great way to combine levelling with some transient control. One of our favourite target sounds for this is bass guitar. Here we have a bass part where the levels are inconsistent but where we also want to use an aggressive compressor to add bite.

2 > Let's tackle the bite first. For an aggressive effect load up a fast design such as a FET compressor. We've chosen an 1176 emulation, setting a fast attack and fast release, with a 4:1 ratio. Adjust the threshold to taste – we're achieving up to 7dB of gain reduction.

3 > Used in this way, the 1176 adds some flavour and edge to the attack, which is precisely what we want. Nevertheless, the bigger picture is that the bass level is still inconsistent. Rather than adjust the 1176 to attempt to achieve this as well, just add another compressor in series after it.

4 > For general levelling, we want a transparent compressor. Try a classic VCA design or a regular DAW compressor (we're using the stock Logic Pro X design). Set the Ratio low (2:1 for us) and the Attack reasonably slow (we've set ours to about 70ms). Ideally you'll have an auto release option, as this delivers program dependent release.

## > Step by step 11. Serial compression *(continued)*

5 > Adjust the Threshold to achieve a few dB of gain reduction (our maximum gain reduction is about 4dB). The reasonably slow attack means this compressor is not grabbing the transients and, combined with the auto release, has a glue and levelling effect. Try bypassing each compressor and then both compressors to assess their impact.

6 > We can also use serial compression on full mixes and complex submixes to combine enhancement and glue. For harmonic enhancement, try valve or transformer-based emulations, such as the Fairchild 670, as these add harmonic distortions even when there is little or no compression.

7 > Load up your compressor across a submix or the master output. We've loaded up the Slate FG-MU valve bus compressor across our mix bus. Set the Threshold to taste. Here we've set the side-chain High pass filter to about 100Hz and adjusted the Threshold for about 1dB of gain reduction. Attack and Release are set to medium.

8 > Now use a second compressor to tighten up the overall mix. We've loaded ours following the valve compressor, though here it's with very moderate compression, the order may not influence things that much. Load your favoured mix bus compressor plugin. We're using the ever reliable SSL Bus Compressor 2.

9 > For mix glue, we're using the 2:1 ratio with a medium to slowish attack (10ms) and Auto Release. With the side-chain high pass filter switched off, the low frequencies are driving the compressor and we've set the Threshold to achieve up to about 3 or 4dB of gain reduction. Overall the mix sounds tighter.

10 > Finally, try switching out both plugins to A/B with the original. This is clearly a more subtle treatment than our first example with the bass. Nevertheless, each compressor is doing its own thing and although the overall metered gain reduction is quite modest, the mix sounds tighter in the low end with enhanced high mid frequencies.

# Mid/side compression

We would typically process stereo submixes and master mixes using regular stereo processors. And in the case of a compressor, although the left and right channels could be processed completely independently, our compressor usually has a stereo linking mode that uses a combined left and right channel sidechain so that the stereo image remains stable.

However, you'll often hear mixing and mastering engineers talk about processing stereo signals using the mid/side technique. This splits the signal into its sum and difference components allowing you to process them independently and then returns them to stereo at the output. Much like recording in MS, this process only requires two audio channels, which means that it's reasonably straightforward to design a stereo compressor that can also process in mid/side (the classic Fairchild 670 for one example).

In software form you get various implementations including controls for both mid and side components. Or sometimes the option to process just one, leaving the other aspect unprocessed (this is the case in TDR Labs' excellent Katelnikov free plugin). Either way, plugins have stereo inputs and outputs so are very straightforward to use.

But why would we choose to compress in mid/side mode? This really boils down to the way contemporary mixes are structured. Often bass-heavy and key punchy elements are in the centre, while the stereo comes from aspects in reverbs, stereo sounds and elements that are panned. Nevertheless, mid/side allows us to reshape the mid components, and apply more suitable compression or even none at all to the side components, and this leads to a different outcome to straight up stereo compression, where the compression is applied to the whole stereo signal. Here we'll run through how to set up and use this technique on a stereo drum mix.

## > Step by step 12. Mid/side compression

1 > Stereo beats are a great starting point for trying out MS compression, and an ideal candidate is a drum kit as by default it typically includes centred aspects such as kick and snare and stereo components in the overheads, room mics and any reverbs that have been added in the mix.

2 > Here we're starting with a multitracked drum kit that combines mono and stereo elements as discussed. That said, we are still processing the overall stereo blend and in this case at the output. We've loaded up a mid/side capable compressor, and on its signal flow page set it to mid/side mode.

3 > Initially we're going to focus on just the mid component. Back in the main compressor window we've used the tab to select the 'mid' parameters and have then selected the opto compressor emulation. The reason we've chosen this is because we can achieve an audible compression that demonstrates the technique.

4 > Whenever you're processing mid and side independently, changes in the balance between these two components influences the width of the stereo image. Compressor automatic gain and make up gain can cause an extra layer of confusion, and we suggest switching this off and using the manual make up gain to manage the balance between mid and side.

## > Step by step 12. Mid/side compression *(continued)*

5 > Now let's set the main compressor parameters. We've chosen aggressive settings with a quite fast attack but slow enough not to crush the transients. With automatic release and a ratio of 5:1 we've set the threshold so that gain reduction is hitting 10dB on the peaks.

6 > This amount of gain reduction has had a big impact on our stereo image, and with manual gain make up, it's time to rebalance the output level. You'll have to do this by ear, as although the meter is telling you -10dB of gain reduction, that's only the peaks. Here we've added about 5dB on the make up control.

7 > Try A/Bing the result to see the difference. The key things for us are the much more snappy, fatter-sounding snare drum and the more solid central image. Also the wide parts of the stereo image remain quite natural. Finally, there's fundamentally more energy in the blend but it's not pumping.

8 > Now let's try processing just the side component. Here we've switched over to our routing matrix and deactivated the mid channel. Then on the main compressor page we've selected the Side tab and chosen the super fast Studio FET emulation, once again deactivating the automatic gain make up.

9 > Our aim is to use the compressor to bring up the stereo aspects of the blend and add some shape to the transients. To that end we've set the attack reasonably fast and the release at its fastest setting. With the ratio at 8:1 we've adjusted the threshold to achieve up to 8dB of gain reduction on the peaks.

10 > As before we've adjusted the make up gain to balance the mid and side (4.0dB for us). Try A/Bing the result. The side component is now more present and has some pump to it. Meanwhile the unprocessed mid component delivers the clean transients. Once again there's more energy to the sound. **cm**

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# Mixing drums with character

## It's time to learn how to use reverb, vintage EQ, compression and saturation to deliver a distinct character to your drums

**> Programming drums using multiple sound sources can result in them sounding incoherent. The solution? Apply a few processing techniques to a drum group.**

In this tutorial, we will discuss the balancing of individual channels and sending all of the drum stems into one group ready for processing. Reverb will be used in order to imprint a distinct ambience and affect the way in which the drums will be perceived.

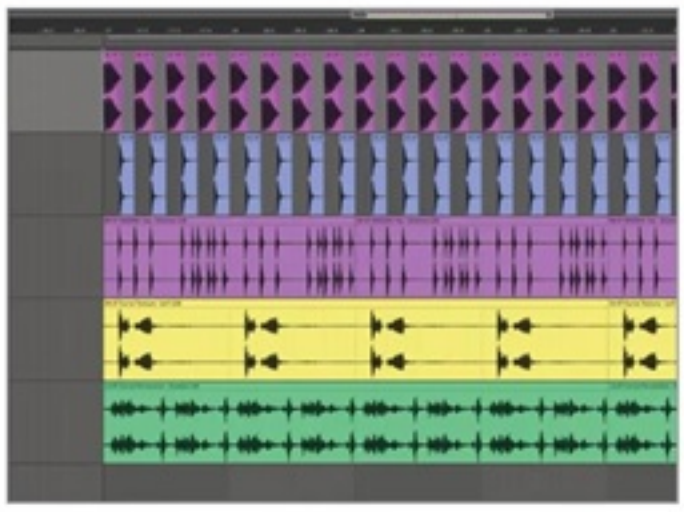
Before modern advancements in digital technology, vintage outboard equipment was used to shape and colour the signal path. Current digital emulations of sought-after vintage gear have come a long way, sounding extremely close to their predecessors. We can apply the same techniques and access a distinct character of the past without breaking the bank.

The main benefit of grouping is the ability to affect all the individual stems with the same processor. This is especially useful when the aim is to accomplish a more unified instrument group. More notably, it's the tools that introduce harmonic distortion are the ones that make the biggest change to the personality of the sound.

After reading this step-by-step guide, we trust you will be armed with the knowledge to add your own signature 'charisma' to your drums. Take this approach and don't be afraid to apply the same principles to other instrument groups.

## >Step by step Consistently characterful drums

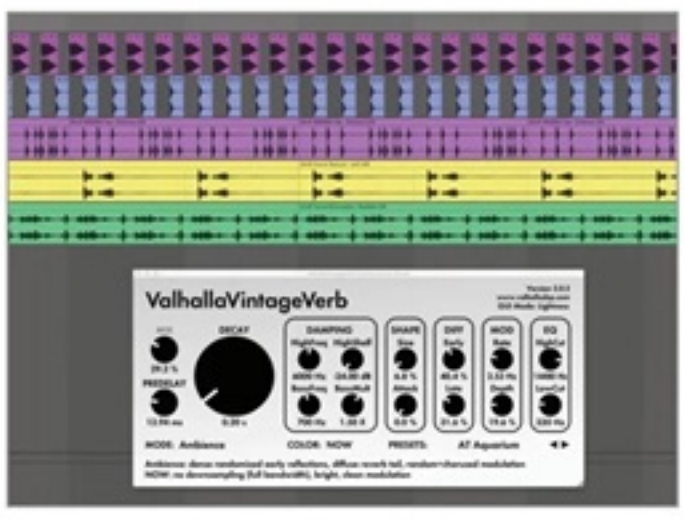
**1** > In the first step we'll programme a techno-inspired drum groove. It consists of the following key elements: kick, hi-hats, drum top loop, low percussion and rumble. The BPM of choice is 128 to keep the energy peak time. All of the sounds are sourced from a variety of Audiotent sample libraries.



**2** > The drums are sounding good, but they lack cohesiveness as each sample comes from a different source. Different equipment and techniques have been deployed to capture the audio. Our aim is to use modern mixing techniques to bring all drum layers together. This results in a unified drum group with its own distinct character.

**3** > Let's start by selecting all of the individual channels and sending them into one group ('Techno Drum Group'). Before we go into the group processing, let's get the mix levels balanced nicely. Otherwise, for example, if you have your kick drum way too loud against the rest of the drums, it will drastically influence how the rest of the processing will react.

**4** > One of the most conventional techniques you'll find to make a group of sounds unified is to apply a reverb. This gives the listener the impression that all of the drums were recorded in the same space. We are not looking for a long, cathedral-like effect, rather a small room which we can dial using early reflections of the reverb.




**5** > Any reverb could work. We used our trusty Valhalla VintageVerb for unprecedented control over the early reflections of the reverb. Keep the decay time short: 20-80ms. We don't want the group to sound unnatural, so we are using a mild amount of the wet signal. We've chosen an Audiotent preset for Valhalla VintageVerb called AT Aquarium.

**6** > Now that the common space is dialled in, let's address the character of the drums using an emulation of a vintage EQ. We're not trying to use a surgical EQ to fix any resonances. The aim is to use broad strokes (low Q values) to gently alter frequency balance. Note that a vintage EQ has a tendency to impart a certain colour which will help unify the drum group.

## PRO TIPS

iZotope Ozone

### GROUP MULTI-BAND COMPRESSION

Adding compression to a bus (group) channel can help to glue it together and create extra energy. Multi-band compression allows you to do the same while giving an extra control over the frequency ranges. You can, say, create energy by heavily compressing the top end and at the same time, gently glue the mid range.

UAD Softube Dyna-mite

### TIGHT PERCUSSION

Percussion samples might have an excess amount of reverb and delay imprinted into the stem. A good processing technique is to use a gate. This will allow you to control the effect tails once they drop below the set threshold, resulting in a tighter sound.

**Ed Strazdas**
**Ed is an experienced dance producer and sound designer, working in the scene since 2004**

**Neil Crockett**
**Sound designer and mixing engineer, Neil is a huge fan of Eurorack modular synths**

## About Audiotent

Our Production Technique experts are from Audiotent, a team of creative producers and sound designers who provide inspirational tools and education for professional artists worldwide. They focus on both techno and progressive genres, offering presets, samples, templates, MIDI files and production tutorials. More from: **audiotent.com**

**7** > The EQ of choice is Universal Audio's emulation of the Pultec EQP-1. We start by selecting a broad bandwidth setting, to give us the smoothest results. Next we add a small 1.5dB boost in the low end at 60Hz. Finally a generous boost of 4dB in the high frequencies at 5kHz.

**8** > Compression can be a powerful tool when seeking a cohesive drum group sound. Firstly, the compression helps to make the transients consistent. Secondly, some vintage compressors, by design, have input or output transformers or tubes that can radically colour the signal. This is more noticeable if the signal path is pushed harder.

**9** > One of our go-to tube compressors is the UAD Fairchild 670. Since our tempo is pretty fast at 128bpm, we choose the fastest setting so that the compressor has a chance to reset after each transient. Using high amounts of gain reduction can really affect the feel of the low-end. So we're compressing moderately, a max of 1 or 2dB of gain reduction.

**10** > If compression introduces an undesired effect for your particular drum group – maybe the transients become too spiky – you could skip using a compressor and apply saturation instead. This will add the harmonic distortion required and level out sharp transients.

**11** > For saturation we use the very flexible Saturn 2 by FabFilter. This is the last stage of the processing and only a gentle touch of colour from the harmonic distortion is needed. Sometimes auditioning different presets can give you instant results. The Light Saturation preset from the Audiotent Saturn expansion bank was a perfect fit for the drum group.

**12** > When programming drums using a variety of sample sources, you run the risk of your drums appearing incoherent. To conquer this problem we can use tools such as reverb, broad vintage EQ, compression and saturation. These tools applied on the whole drum group will imprint a unique sonic character as well as making your drums sound unified. **cm**

THE ESSENTIAL GUIDE TO

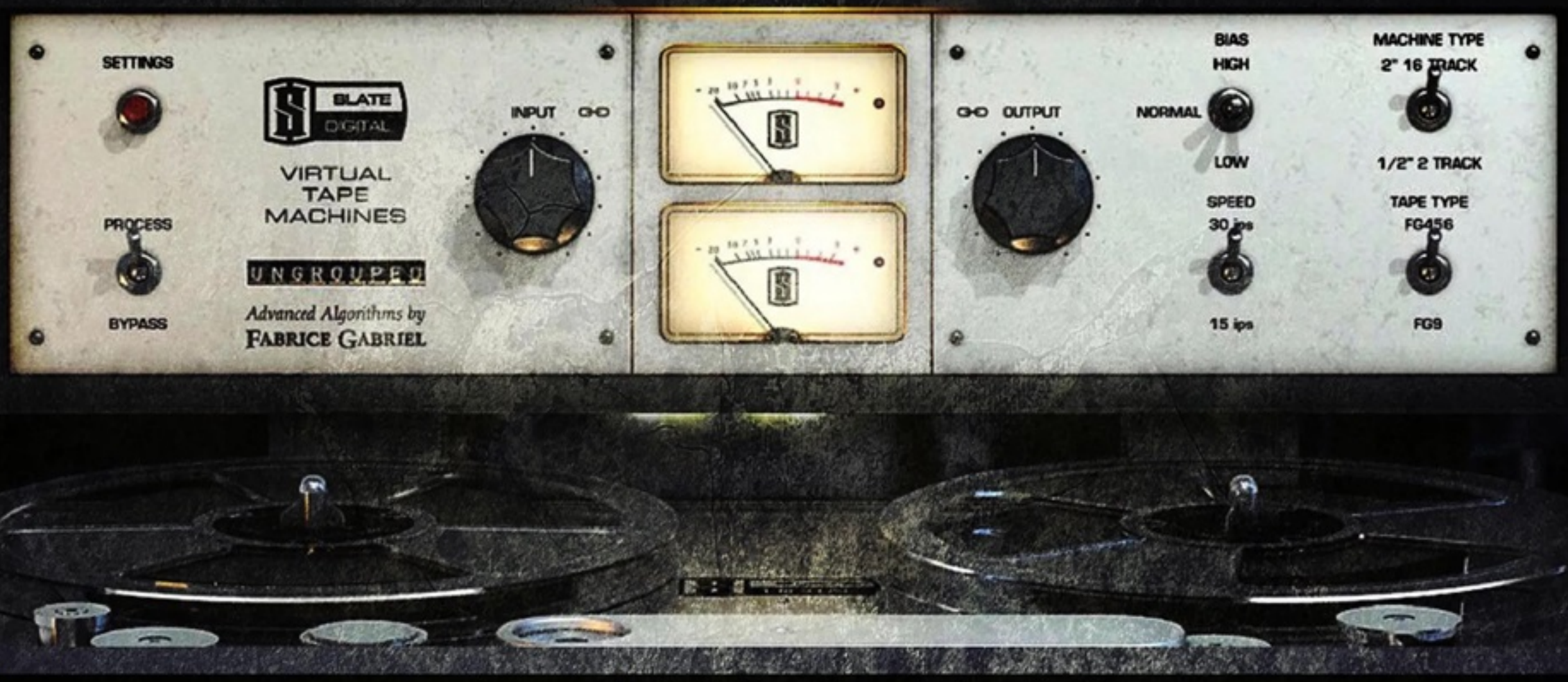
# ANALOGUE DIRT

Your digital mixes may well be too perfect! Here's our guide to bringing some humanity back into your productions and how to get deep, down and dirty with some 'analogue' processing

**If you're old enough to remember the 1980s, you'll recall it was a time when 'digital' became the cool, new thing. And not just in music production. Everything went digital, from clocks to watches, from cash tills to calculators. And in music making we got the skips in to take our compressors and great big modular synths away! In came digital synths, computers, hard disc recording and endless ones and zeros. Forget those silly knobs and dials kids – we had endless menus to step through and could now record in binary! Forget tape, hiss, noise and wobble – clean audio perfection was just round the corner!**

How fashions change. Somewhere along the march of digital, we decided that it was a bit too clean and harsh. And now, of course, analogue is back! Modular synths – many analogue in nature – are all the rage and those digital, menu-driven synths of the 80s have been consigned to their deserved place in history. But we're not just talking synths. The actual processing of sound also became too clean and clinical. We now yearn for vibe, warmth, presence and authenticity. But the analogue gear used to produce this – synths, vintage beatboxes, room-sized mixing consoles, and whirring tape machines – now cost an arm and a leg. Besides, we're *Computer Music*, right? And now you can use your DAWs to combine digital clarity with analogue charm, realism and punch, and that's what we'll show you in this month's cover feature. We'll explain just what 'that analogue sound' is, and reveal some easy techniques to add the right amount of it to your mixes. We'll show you how to use the best analogue-style plugins, and how classic mixing methodology can help get the most from them. Best of all we'll do most of it using plugins you already have by way of the mighty *Computer Music* Plugin Suite.

So it's time to defrost your sound and bring some humanity back. Let's fire up our DAWs and return some classic warmth to your mixes.



# Why analogue is better

**So we know that digital processing can be too perfect or harsh. And by comparison, analogue is usually described as 'warm', 'punchy' and 'fat'. But why is this?**

Digital is a practically perfect system for representing sound. For 44,100Hz audio, for example, all frequencies up to half that point (22,050Hz in this case) are perfectly represented. With audio processed at 64-bit and saved to 24-bit files, signals can be super-loud or incredibly quiet with no practical loss of quality. When digital audio is copied, routed, processed or adjusted in level, the quality remains intact. No noise is added. No frequencies are removed. Dynamics and transients are unaffected. It's a near-perfect medium.

Now let's look at a typical analogue signal chain – see the diagram on p16. Imagine that an analogue synth is producing a basic wave. Its components are not perfect, so the source waveform is not an exact saw to start with. Then the analogue circuitry adds noise which is caused mainly by the constant motion of electrons in the equipment. It's

Every component in an old-school analogue setup would add its own imperfections to the final signal

"All analogue circuitry adds background noise ... there's no way around it"

basic physics, and there's no way around it. So the output of the synth will already have a little noise, and the electronics in each successive piece of equipment will add to this. Likewise, some analogue gear can pick up hum from the current of your mains power supply – 50Hz in the UK and Europe, 60Hz in the US. Each circuit may also affect frequency content, such as rolling off treble or adding a bass hump. In addition, all analogue circuitry adds subtle distortion and even phase distortion. These are subtle changes, but they all add up.

The medium of tape adds further hiss, frequency bumps, distortion and dynamic squashing. Tape speed varies slightly but constantly, causing tiny timing/pitch wobbles called wow (slow) and flutter (rapid).

Now factor in drum machines, microphones, and analogue effects, and their associated signal paths. Each brings more dirt, hiss, hum and distortion. If you run out of tape tracks, you'd have to bounce them down to a spare track, adding another layer of permanent hiss and tape grit.

With all that going on, you can begin to understand why analogue sounds the way it does – just imagine everything our synth's saw wave has been through by the time it finally hits the speakers. In contrast, in your DAW, you can synthesise a saw wave that's perfectly preserved all the way to the final output. It might be 'better' on paper, but in practice, a mathematically ideal saw wave isn't always pleasing to listen to.

The good news is that there *isn't* an intrinsic reason why digital can't sound like analogue – we just need to take steps to intentionally make it sound that way.

In your DAW – or the free *Computer Music* Plugin Suite – you have the tools to do this. You have everything you need to replicate all the additional elements that analogue processing brings in.

Over the next few pages, then, we'll show you how the analogue signal path can be replicated using common DAW plugins. We'll then show you how you can use your free *Computer Music* Plugin Suite plugins to emulate a series of common analogue processes. First, though, let's look at a typical analogue recording scenario.

# The evolution of distortion

Professional audio gear is designed to give acceptable noise and minimal distortion at appropriate signal levels. By keeping levels throughout a real signal path around 0dBU, the producer should get good results.

Even so, producers were desperate to keep noise levels down, so they would often intentionally run signals 'hot' to increase the signal-to-noise ratio. Most analogue gear breaks up and distorts in a pleasant way, and this characteristic eventually became associated with classic production.

So what *is* distortion? At its simplest, when a signal is played too loud, the excess is chopped off or 'clipped'. A sine wave's smooth top and bottom would be flattened off, making it more like a square wave. This process introduces harmonics, in the same way that 'squaring off' a sine wave does.

Analogue distortion comes in a variety of flavours and characters. Early electronic gear used valves (aka tubes), which are associated with producing even and odd harmonics. The 1950s saw valves superseded by the more compact solid-state transistors, which emphasise odd harmonics when distorted, sounding sharper and more aggressive. Valves aren't used in consumer electronics anymore, but they still have a place in guitar amps, preamps, and vintage-flavoured effects. Tape lies somewhere between valve and transistor, producing stronger odd harmonics.

While distortion is a technically accurate term, we often use the terms saturation or (over)drive instead, referring to mild and moderate amounts. In music production, 'distortion' is usually an aggressive effect – think metal guitars and gabber kick drums.

## The classic analogue recording signal path

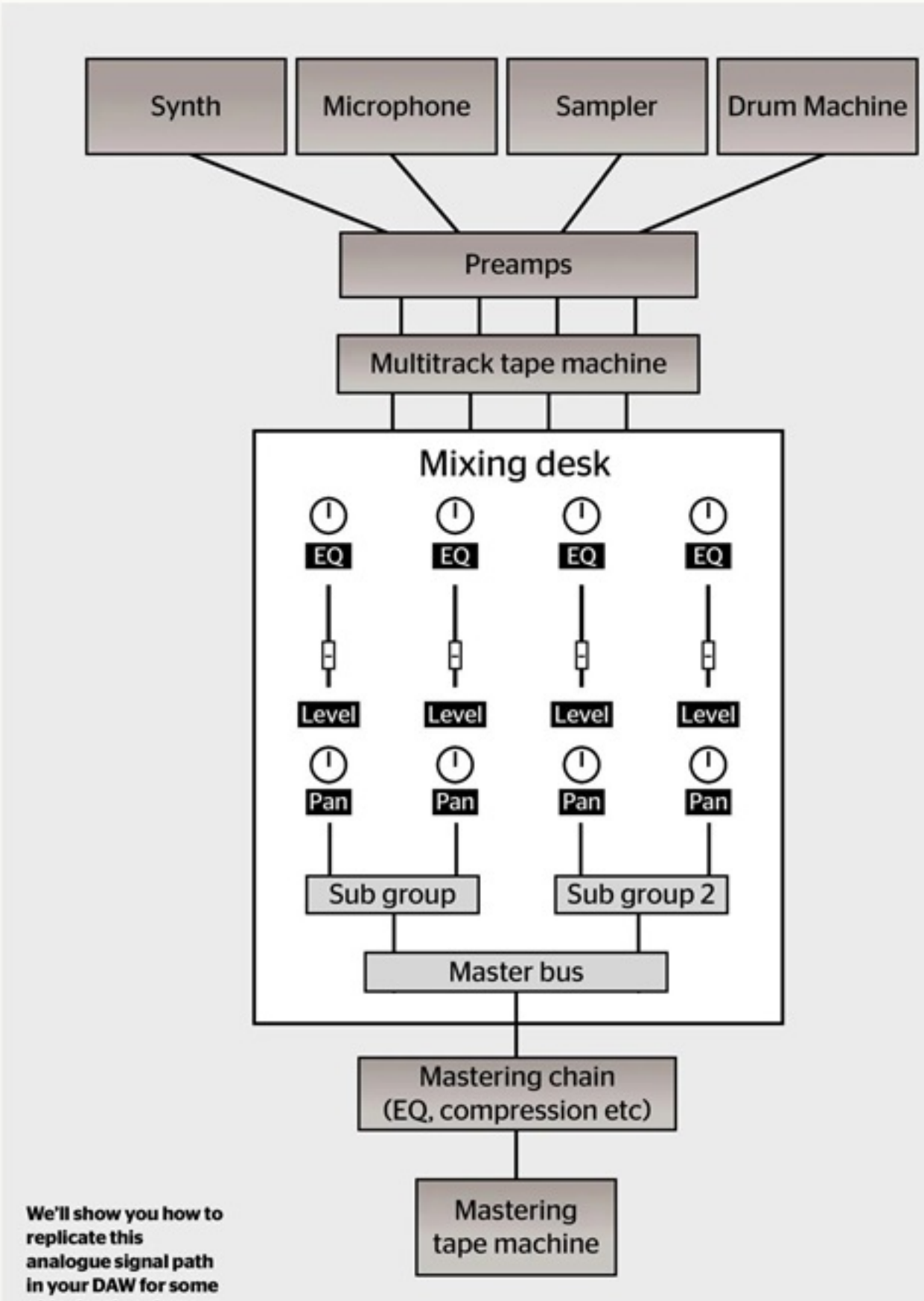
**Before we go about emulating the quirks of electrical circuits in our DAW, let's take a look at the original signal path we're trying to emulate. The diagram on the right shows how such a classic setup works. Let's start from the top and work down...**

**A multitrack tape machine would add yet more saturation, and inconsistent tape movement would lead to very slight pitch and timing changes. Analogue mixing desks would introduce saturation and crosstalk (ie, signal bleed between channels). With the number of complex connections made here, electrical noise is a likely addition, and there are plugins designed to replicate this on every channel of a mix, as well as the master bus.**

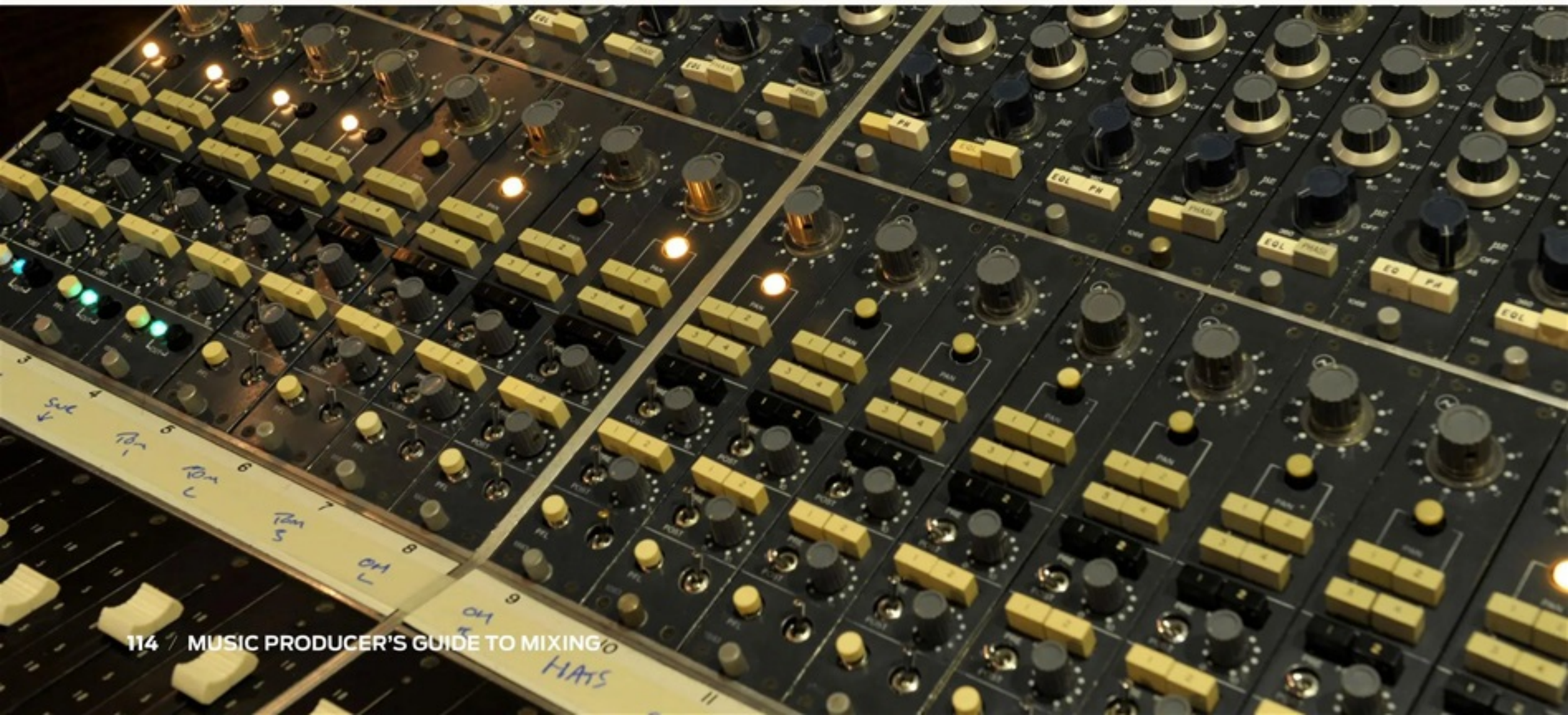
**By using plugins to emulate a signal path like this, you'll dial in the cumulative warmth and cohesion of the analogue studio. In the tutorial below, we'll show you how to easily add authentic vintage vibes to a whole track using common software tools.**

**Om the opposite page we'll go back to basics with 'analogue' as we explain how to replicate the entire signal path in your DAW. We'll show you how you can emulate every stage, from tape through mixing console through effects processing, all using common plugins.**

**Then from page 123, we'll show you how to reproduce some common analogue processing tasks, all using your free plugins from the (mighty) *Computer Music* Plugin Suite.**




**We'll show you how to replicate this analogue signal path in your DAW for some old school analogue warmth and dirt**



# The Pro Producer's Guide To Analogue

Getting a distorted analogue sound is a great way to add filth to digital productions. Here's a look at the classic analogue signal chain and how to recreate it in your DAW

**> It goes without saying that the ability to record and edit audio in the digital domain brings enormous benefits, enabling us to produce punchy, cutting-edge music entirely within our computers. However, there's still something undeniably special about audio processed in the analogue domain. Wouldn't it be great, then, if you could make your digital recordings sound more analogue?**

Thanks to an increasing number of extremely talented plugin developers, this has very much become a viable possibility, as they painstakingly model the gloriously chaotic analogue processes behind magnetic tape, valves, transformers and more. Of course, all of these things have been around in plugin form for ages now, but the current generation focuses particularly on the subtleties and nuances, emulating the effects of tape and desk circuits used at regular operating levels.

In general terms, the analogue recording chain runs from a mic or DI, via a preamp, desk channel and sometimes desk bussing, out to a tape deck. The resultant multitrack recording is then replayed via desk channels and summed to stereo. Within this process, there's an awful lot going on that simply doesn't factor into a DAW setup.

Probably the most significant element is the tape machine, where various pre-emphasis and playback EQ circuitry combines with your chosen operating level, biasing and, of course, tape type to greatly influence the outcome.

Beyond this, desk bussing also influences sound, and when you punctuate the signal chain with transformer and amplification circuits, some of which may be valve-based, there are clearly many sources of analogue colouration being brought to bear.

In these walkthroughs we'll put some of these tools to the test by taking some pristine digital sounds and analogue-ising them. We'll apply our plugins on an 'as needed' basis, using tape, transformer and/or valve processing on individual sounds or groups of tracks where they're best suited. Don't forget to check out the accompanying audio examples.

# Drum and tape – a match made in heaven

It's often said that drums and tape make the perfect pairing, but why? And is this still a meaningful suggestion in the digital age?

If you've ever recorded drums on a high-quality professional multitrack tape machine, the experience of playing them back and hearing the sound change can be both exciting and unnerving. Tape typically applies a level of mid- to high-frequency smoothness, overall glue and subtle compression to the sound. However, the degree to which this occurs is dependent on many small factors, and it can be pretty baffling knowing how to use it to your advantage.

The factors involved include the level captured to tape, the tape type, tape speed (and the inherent head bump or bass boost associated with it), the choice of EQ curve and, of course, the type of machine used. And to make matters even more complicated, a tape machine has two heads – sync and repro – which sound slightly different to each other. The first can record and play back, with heads physically lined up to enable overdubs to be carried out perfectly in time; the second offers a higher-quality 'playback-only' solution for when you get to the mixing stage. Typically, you'll be listening on the sync head when tracking and making recording decisions.

> Tape affects an audio signal in a beneficial way, but undesirable changes may arise

Broadly speaking, analogue tape affects an audio signal in a beneficial way, but there are some changes that may arise which are less desirable, including crosstalk, print-through, edge tracks, softening of transients and tape hiss. The last of these can become quite significant if you EQ high frequencies heavily after recording. Thankfully, the plugin emulations we're using in this tutorial focus on the positives, so although you might be able to dial in some tape noise if you want it, you can thankfully switch it out or turn it down if it's becoming a problem.

In this walkthrough, we'll put the emulated tape/drum combination to the test. Unlike in the analogue world, we can adjust our drum tracks individually, deciding how much tape effect is applied to each. We can also adjust the tape type and line-up individually.

## > Step by step

### 1. Tape emulation

**1** > There are quite a few tape emulation plugins to choose from – for this walkthrough we're going for Universal Audio's Studer 800. Not only is this a meticulous virtualisation, but the A800 is an old engineer favourite for drum tracking. With its many presets, this plugin allows us to very easily replicate a typical drum tracking situation.

**2** > Tape type plays an important role in how noisy things are and how much saturation you can achieve, and some of the more recent high-output tapes can be much harder to saturate. Here we're going for an older but very popular tape for drum tracking: Ampex 456. This saturates easily and should work well for the kick and snare.

**3** > Let's start by loading up a very linear preset so that we can hear what, if any, difference a linear machine makes. We're also going for a tape speed of 15ips. Although more noisy, this imparts a slightly different frequency response with better low end. To allow for peaks, tweak the input level so the VU is reading between -5dB and -10dB. (Audio: Step 3a and Step 3b)

**4** > With our linear setting, we're getting some enhancement to the sound. Now let's see if we can improve things by pushing the level to tape. Initially this adds to the sound, enhancing the thump of the kick, but eventually the returns diminish and it starts to sucks the life out of the sound. Back the level off to the optimum point.

You can get the sound of the SSL 4000's legendary buss compressor in your DAW via numerous plugins, including Slate Digital's Virtual Console Collection

## > Step by step 1. Tape emulation (continued)

5 > We have some other parameters to play with. We can switch to the Sync head or, indeed, just the Input. Both are worth trying, with the latter modelling only the input electronics. There's also the Noise setting, which introduces both mains hum and noise based on the line-up preset (CCIR or NAB). We keep the noise but turn down the hum.

6 > For our snare drum we're going to stick with the 456 and 15ips, once again starting with the same linear preset as the kick. This time, however, more drive is the order of the day. As a final twist, we can adjust the pre-emphasis record EQ setting to change the high-frequency response.

7 > For our overheads, we go for a different approach. Using a quieter and more linear tape (GP9) at 30ips, again our starting point is a linear preset – but note that the operating level is now higher with the high-output tape. A/B-ing the sound for the overheads reveals the addition of some top-end smoothness. (Audio: Step 7a and Step 7b.)

8 > To step things up, we drive the tape harder by increasing the Input level. This adds a little crunch and some 'glue' to the sound. For our final trick, we go in and try overbiasing the tape. This increases saturation and reduces the high-frequency output, which is most noticeable on the hi-hats and snare.

# Desk matters – virtual consoles and bussing plugins

If you're happy with the process of emulating classic tape machines, the thought of emulating analogue mixing desks shouldn't be at all intimidating. In some respects, though, the whys and wherefores of a desk's behaviour are hard to quantify. Analysing and emulating the channels and output section individually is one thing, but creating a full-on model of how it all combines, blending those channels depending on signal levels and frequencies, couldn't be harder. And that's before you get into the various operating modes and specific signal paths of each individual desk.

Desk plugins tend to offer separate channel and buss versions, the idea being that you insert these on the first insert point of each channel or buss respectively. Although this is not an exhaustive list, some great examples include Slate Digital's Virtual Console Collection (VCC and part of Virtual Mix Rack), Waves' NLS Non-Linear Summer or Scheps Omni Channel, Sonimus Satson, the Brainworx bx_console series. Not all of these plugins actually name the mixing desks they model, but most do allude to them, so you can be reasonably certain of what you're using, and that it will sound great.

> The whys and wherefores of a desk's behaviour are hard to quantify

Satson is the simplest, offering one desk option (essentially an SSL 4000 Series). VCC – which is two plugins, Virtual Channel and Virtual Mixbuss – now has six desk options, including SSL, Neve, API and Trident models. Brainworx bx_console offers two SSL emulations plus a model of a rare Neve VXS console. Given that this is owned by Dirk Ulrich, the CEO of Brainworx, you'd expect to hear the company's famous modelling in full effect! Finally, Waves NLS is three desks owned by three particular producers: Mike Hedges' EMI TG, Yoad Nevo's Neve 5116, and Mark Stent's SSL 4000G.

Beyond all this, some plugins offer crosstalk for stereo instances, optional automatic gain control and ganged grouping for easy multichannel adjustment. Some of the extra features with VCC include level calibration, meter ballistics and oversampling settings, while Waves NLS includes more than 100 discretely modelled channels for the collection.

## > Step by step 2. Console emulation

1 > We'll use the channel and buss plugins from Slate Digital's VCC – part of the company's Virtual Mix Rack – as they sit at the more subtle end of the desk emulation spectrum. It's best to start out mixing with them in place, so the first step is to load them into VMR and then the first insert slots of the channels and the master buss.

2 > Virtual Mix Rack offers an almost insane amount of processing in one rack (pretty much all of Steven Slate's plugins) so you can – should you wish – have multiple classic consoles applied to one channel or (here) the same console applied to two channels plus other outboard loaded into VMR.

3 > Now let's try the various desk emulations. In this instance we're going for the upfront API desk (USA A from the six presets) on our drums, with the Neve on everything else. On our master buss we opt for the classic SSL mix buss (Brit4k). Ah, the awesomeness of being able to combine desks.

4 > Finally, it's drive time. By using a group Drive control, we can adjust different sets of sounds very quickly. The Drive control on VCC simply changes the non-linear response without affecting the balance. Having ramped up the channel Drive, we reduce the Drive slightly on the master buss.

# Other ways to get that 'analogue' sound

As we've already noted, the analogue recording path is pretty complex, with plenty of points at which noise, distortion and non-linearities can be introduced. So, once you've got things going with some tape and desk emulations, there are still various other things you can do to add further analogue flavour.

For warmth and a good balance of harmonic distortions (see *Harmonic Distortions in the Analogue Recording Chain*, below), look for both valve- and transformer-based plugins. For more edgy sounds, seek out solid-state designs, and don't overlook emulations of classic outboard gear – compressors and EQs, mainly.

As CPU power has increased, so have the ambitions of plugin designers in terms of accurately modelling all aspects of the analogue path. So, if you know that a unit has a valve amp section (the Pultec EQP-1A, say), check to see if the designer has included this in their plugin emulation of it. The other thing to bear in mind is that this process is additive, in the sense that making small changes to many tracks of audio will have a cumulative effect, so don't go hunting for massively coloured processors when only subtle changes are called for.

**Slate Digital's Virtual Buss Compressors can be loaded as a rack, shown here, or individual plugins**

## Plug away

As with the tape and desk emulations covered on the previous pages, this is by no means an exhaustive list, but here are a few suggestions to get you started.

For valve sounds, Wave Arts Tube Saturator is a classic, free Mac plugin. Based on a 12AX7 valve, it models the two triodes in the 12AX7 to emulate two cathode triode amplifiers, much like the preamp stage of a tube hi-fi amp. At the lowest drive level, it's pretty clean with just a little harmonic distortion.

Next, SPL's Twin Tube, another classic plugin. Based on their hardware unit of the same name, this one includes separate saturation and harmonics circuits. The first creates harmonic distortion as you push it into distortion, while the latter adds harmonics at one of four specific frequencies (2, 3, 6 and 10kHz). Combining the two helps to fatten sounds and add air.

Although the name implies a level of aggressiveness, SoundToys Decapitator can also be used in subtle ways. It includes five different flavours based on five specific valve and transistor preamps. Of course, if you do want to ramp it up, all five can deliver that too.

> For warmth and balance of harmonic distortions, seek both valve and transformer based plugins

While we're on the subject of ramping it up, iZotope's Trash is designed for audible distortion, but can be tamed via its waveshaper's curve editor if you want something more subtle. Mellowmuse SATV, meanwhile, boasts four analogue options – transistor, transformer, tube and tape – each with individual drive, mix blend and grouping.

Another oldy in terms of its longevity (and, yes, looks) but a very good performer is Voxengo's VariSaturator, which includes both valve and digital saturation, with split processing between high and low bands.

Finally, we have to mention Slate Digital again and their Virtual Buss Compressors. With extensive modelling of analogue non-linear behaviour, there's a certain 'black box' nature to these plugins that you have to experience to fully appreciate.

In fact the company's Virtual Mix Rack is one of the most elegant ways of combining consoles, outboard, effects, you name it, in one simple solution – see our walkthrough on the previous page for more on this.

## Harmonic distortions in the analogue recording chain

When we talk about the benefits gained from an analogue signal path, it's the harmonic distortions that have the most obvious sonic effect. These are the harmonics that are generated by the circuit and added to the fundamental frequency, of which they're multiples, numbered sequentially. So the second harmonic is twice the fundamental, the third harmonic three times, and so on. Viewing these from a pitch perspective, it's only when you hit the seventh harmonic and beyond that the multiples start to become dissonant against the fundamental. Helpfully, though, the amplitudes of these harmonics tail off anyway.

When it comes to our analogue components, harmonic distortions increase as each component is driven, and it's their relative levels (odd versus even) that influence the overall sound.

Typically, the even harmonic distortions (second, fourth and so on) sound solid and complementary, fattening up the fundamental, while odd harmonics sound more complex and edgy, enhancing the fundamental. Transistor circuits generate predominantly odd harmonics, while transformers produce both odd and even.

Magnetic tape recorders also produce both odd and even harmonics, although the even ones tend to be less pronounced than those produced by a transformer. Finally, valves typically put out a mixture of odd and even harmonics, although the onset of saturation tends to be gradual – hence the popularity of valves for driven sounds.

## > Step by step

## 3. Emulating other types of analogue processing

**1** > Our bass is solely a DI signal, and although we like the roundness, it lacks character. Adding tube saturation by way of Wave Arts' free Tube Saturator, is a great way to mimic a decent amp, without having to mic it. We get some effect at very low drive, but opt for a bit more drive and some extra saturation. (Audio: Step 1a and Step 1b)

**2** > Our acoustic guitar sound has top-end sparkle and body but seems to lack bite, sounding a bit weedy. By using a combination of saturation and enhancement, courtesy of SPL's Twin Tube, we can bring back its presence. Here we're dialling in the harmonics at 6kHz.

**3** > Saturation doesn't come much better than by way of Soundtoys' Decapitator. In truth we can apply it to most parts of our mix to great effect – and are tempted to do just that. But as we say in the main text, a little can often go a long way, so try applying some mid range saturation to the electric guitar.

**4** > Our snare sounds a little too polite, and although we don't go for transistor distortion often, here it could be just the thing. First, we dial in lots of drive to get the desired effect, then use the handy Mix control on Mellowmuse's SATV to blend in the desired balance. (Audio: Step 4a and Step 4b)

**5** > Rather than head for EQ on our kick drum, we're going to use a combination of valve and solid-state saturation, courtesy of Voxengo's VariSaturator. We've positioned the crossover at 1kHz, using valve saturation on the low frequencies and solid-state on the high frequencies. The result is a more punchy kick.

**6** > The piano sound is pretty well balanced but lacks air and feels a bit congested in the low mids. With an emulation plugin such as Slate's FG-Mu, we can take advantage of its analogue flavour without actually compressing the signal at all. The result is a more open and engaging piano sound. (Audio: Step 6a and Step 6b)

# Bringing it all together

Back in the days of analogue, equipment choices were less flexible than today – but as is the case in many areas of the digital world, working in the software domain offers the musician almost too much choice. Keep your head, however, and this is actually a good thing, as there are plenty of sounds – particularly quiet or very dynamic ones – that really don't benefit from being recorded to analogue tape.

Noise, for example, was a big issue with classical recordings, as the music's dynamic range really pushed the capabilities of analogue tape. It's no surprise, then, that that particular sector of the industry shifted to digital recording technologies as soon as it could. You might also have noticed that classic analogue desks, such as those by SSL and Neve, incorporated gate/expanders on every channel, with a view to keeping tape hiss and other system noise at bay.

In the days of analogue, equipment choices were less flexible than today's

One of the most appealing benefits of plugin emulations, then, is that they not only put us in control of the sound with far more precision than the original gear, but also that they let us apply it 'as needed'. This extends to controlling parameters in far more detail than is possible in the real world, including things like noise and hum, as well as tape levels, tape biasing and drive levels in general.

In the walkthrough below we're going to bring everything together, taking our multitrack of sounds and processing each element using what we feel are the most appropriate techniques. Along the way we'll A/B the results to hear what difference each change makes. We'll use our previous examples as a starting point but stay open to adjusting these sounds as needed – in essence, we'll do whatever works best for the overall picture.

Finally, we'll look at the master buss. Although mastering is still the final stage of production, any mix engineer worth their salt will tell you that they're already making pre-mastering decisions at the mix stage, using stereo compression and EQ, and, if available, mixing to half-inch tape. So, our final stage will make use of a mastering tape machine for some added analogue flavour.

## > Step by step 4. Combining analogue emulation techniques

1 > For our drums, we've gone for a favourite: Ampex 456 at 15ips by way of the UA Studer 800 plugin. By ganging the plugins together we're driving the tape hard, which adds presence to the sound. The kick is too squashed, so we ease back its input level into the plugin, turning up its output to compensate. (Audio: Step 1a and Step 1b)

2 > Now let's add some bussing. We position these plugins straight after the tape machine to emulate an analogue signal path. Once again, grouping them makes things simple as we can increase the drive across them easily. The Slate VCC Neve desk is working wonders on the bottom end of the kick and snare, so we go with that.

3 > Rather than head for a bass compressor, we load up a modern rock-style setup for the tape. By driving this, we get just the right containment for the bass. Then we load a plugin that we know emulates the valve amp section – the IK Multimedia EQP-1A. This gives just a touch of upper-mid enhancement.

4 > To give the piano a bit more flavour, we reach for a valve processor. The key here is to add character without it ever sounding distorted. The right tool for the job is Wave Arts' free Tube Saturator, which brings the sound together a bit and adds some edge, making the piano sound less clinical. (Audio: Step 4a and Step 4b)

Although your Mac or PC offers infinitely more recording and mixing power than this lot, there's still an awful lot to be said for that lush analogue sound

## > Step by step 4. Combining analogue emulation techniques *(continued)*

**5 >** For the electric guitar part, we want more edge and some mid-range crunch. Here you can experiment with your stock DAW plugins like Phat FX from Logic, Magneto in Cubase or Ableton Saturator in Live.

**6 >** We're taking a slightly different approach for the acoustic guitar, with just a desk emulation. Driven pretty hard, the Trident-style desk from the Slate Digital VCC adds brightness and presence to the sound. (Audio: Step 6a and Step 6b)

**7 >** Moving on to the master buss, we try a trick favoured by Tom Lord-Alge, using FG-Red from Slate Digital's Virtual Buss Compressors trio of plugins – but only for its transformer flavour. Thankfully, with its dedicated control, we can add as much or as little of this as we like.

**8 >** Our final dollop of analogue glue comes courtesy of the half-inch machine. This time we're after the most hi-fi option, so we go for high-output tape – FG9 – at 30ips with Normal Bias. Note that the line-up is not enhanced and we're not driving the tape hard. (Audio: Step 8 – whole mix before/after processing)

> Step by step

## 5. The common distortion types and how they're used

1 > A huge part of the analogue sound is really all about distortion. Let's take an audio tour of its many flavours and varieties. Tape saturation combines transient smoothing, compression and soft signal distortion to progressively glue and fatten. Emulations often apply EQ and simulated tape hiss and flutter. It's great for warmth and 'glue', and can be applied to every track.

2 > Like tape, valve saturation progressively adds harmonics in a musically pleasing way, with distinct colouration. Triode valves generate largely even harmonics, while pentode circuits give odd ones, often considered less musical, but deeper and richer. Saturation is usually used subtly, rather than pushed to the point of obvious distortion – though it's always an option!

3 > Many types of distortion employ clipping behaviour, but when we talk about clipping in music production, we're usually referring to basic hard and soft clipping. While pure digital clipping can sound harsh, a dedicated soft/hard clipper can reduce peaks invisibly when used moderately. In the mix, it's useful for keeping signal levels in check, and great for transient-heavy sources.

4 > Guitar overdrive pedals like the Ibanez Tube Screamer apply mild distortion, often coupled with EQ/filtering and a level boost, all designed to overload the input of the amplifier they're connected to, adding aggression to a sound which could already be quite 'hot'. Heard in isolation, it has a basic, crunchy timbre that retains dynamics – halfway between the subtlety of saturation and all-out distortion.

5 > Fuzz is basically very strong clipping, usually used as a guitar effect. It adds harmonic distortion and massive sustain, and obliterates dynamics. A filter to roll off the treble can make it more versatile. One of the all-time classic fuzz boxes, Electro-Harmonix's Big Muff – of which Universal Audio's Bermuda Triangle is a great emulation – has just three controls: Sustain, (output) Volume and Tone (filter).

6 > Confusingly, there is a distortion type called 'distortion', derived from the world of guitar pedals. Most associated with heavy guitar music like punk and metal, distortion may incorporate multiple EQ, filtering and clipping stages, arranged and tuned to give extreme results while retaining articulation. Unlike overdrive, distortion pedals are intended to largely create the distorted tone by themselves.

7 > Guitar distortion heard on its own can sound very fizzy and harsh. The guitar sounds we're used to hearing come from a microphone placed in front of a speaker cabinet in a real room, all three of which heavily colour the frequency response. So, for an authentic tone, these aspects must be simulated too, and many plugins offer this. We can switch the cabinet simulation on and off to hear the difference it makes.

8 > Bit-depth reduction gives us the crunchy sound of old-school sampling and vintage home computers. It adds grit and edge, which can help sounds cut through busy mixes. It also adds a hissing sound, noticeable on signals like reverb tails, so you might want to gate signals before bitcrushing, and to apply reverb afterwards. Unless that's the old-school, sampled vibe you're after!

9 > Sample-rate reduction decreases the temporal 'resolution' of the digital signal. The most recognisable result is a kind of atonal ringing with a metallic edge. You can roughly tune this ringing by changing the sample rate, or modulate it for distinctive riffs and digital effects. Modulation of the sample rate is often used as a sound effect in modern sci-fi soundtracks.

# Level, EQ and distortion

When working with distortion processors, it's vital to consider your levels and gain staging at both the input and output stages. When we look at the basic relationship between gain and distortion it's simple: raise a signal's gain into a distortion plugin (by turning up its output, using the distortion device's 'input gain' parameter, or a combination of both), and the waveform will be clipped, shaped or otherwise altered. At the other end, the distorted signal that emerges will typically sound far louder than the undistorted signal.

In practice, then, there are two key points for the producer to bear in mind: firstly, the level of your input (ie, your pre-distortion signal) will greatly affect the distortion response; and secondly, you'll almost always need to turn your post-distortion signal down a fair bit to match the level of the unprocessed signal – this is why most distortion plugins feature a bypass button and output gain/trim parameters.

Just as overall level impacts the distortion response at both the input and output stages, so

**You can affect particular frequencies to control the distortion response**

do the levels of individual frequencies. By EQing a signal before it enters the distortion device, you can affect particular frequencies to control the distortion response, and control which frequencies the added harmonics are generated from. This can be used in a corrective sense – if a particular frequency in the source signal is generating nastiness in the distortion, for example – or for creative uses, such as pushing specific frequencies into the drive stage.

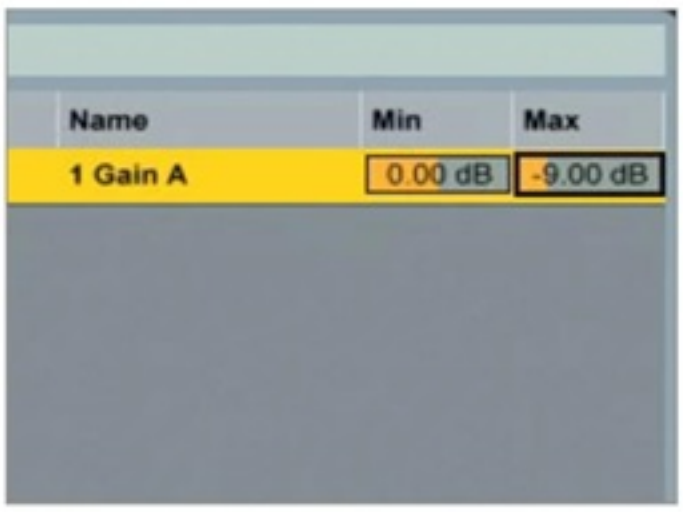
On the flip side, EQing post-distortion is a handy way to 'clean up' emphasised frequencies, which is often useful in a corrective sense, subduing any unpleasantness the distortion has introduced. Combined with pre-EQ distortion, a vast array of timbres can be coaxed from even the simplest clipper.

## > Step by step 6. EQ and distortion

1 > Let's explore the relationship between pre- and post-distortion EQ. Load BassGuitLoop.wav on an audio track in a new 110bpm project. We're using Live as our host, but you can follow the principles in any DAW. Insert Graffio CM, turn off the Exciter and Bit Reducer modules, then crank the Saturator's Gain knob to 25dB, and pull the output Level knob back to -11.5dB. We can improve this fuzzy effect with a combination of pre- and post-EQ.

2 > Load an instance of Live's EQ Eight device, being sure to place it before Graffio CM in the chain. On the EQ's analyser, we can see that the majority of the loop's energy is focused around the low and low-mid regions. It figures that it's these frequencies that are pushing hardest into the distortion.

3 > Highlight both EQ Eight and Graffio CM in the chain, then hit Ctrl/Cmd-G to group them into an Audio Effect Rack. Let's set up the EQ so it will pull down these heavy low frequencies: activate EQ Eight's first band, leave it set to a low shelf, then set the band's Frequency to 120Hz and Q to 0.5.




4 > By assigning the level of our low shelf cut to a single macro knob, we'll be able to control other parameters with the same macro. Right-click band 1's Gain knob and select Map To Macro 1, then head into the Rack's Macro Mapping editor and set this parameter's Minimum to 0dB and its Maximum to -9dB.

5 > Now the EQ's low shelf band is pulled down when we turn up Macro 1 – and we can use this same macro to apply an opposing low-shelf boost after the distortion plugin. Duplicate the EQ Eight instance and place this copy *after* Graffio CM in the chain (within the Rack), then open the Macro Mapping Editor and set this EQ's Minimum to 0dB and Maximum to 9dB.

6 > Now, turning up Macro 1 will simultaneously pull down a low shelf cut pre-distortion, and boost the same frequencies post-distortion. This tames the heavy low frequencies entering the distortion stage and boosts them back again after, retaining their weight and smoothing out the effect. To finish, head into Graffio CM and adjust settings to taste to tailor the response to your liking.

## > Step by step 7. Emulating analogue gear with filtering and EQ

1 > It can be tempting to ladle on every last analogue-modelled plugin you've got in the quest for a warm and punchy mixdown... but you can push your mix in that direction with simple EQ and filtering tricks inspired by analogue gear. Start by loading the WAVs named 'ProgFunk...' on audio tracks in a 132bpm project, then loop them (this demo isn't in 4/4 time).

2 > The drums in this mix have too much treble information, so add a Satson CM to the Drums track and set Low Pass to 7 to smooth away the top-end digital harshness. Set High Pass to 2 to tighten the low end, controlling the excess sub flab. Satson CM's gentle 6dB filters are perfect for these tasks, and we'll be using them subtly throughout the mix.

3 > The drums are more focused, but we'd like to unify the bottom end with the bass guitar. Using graphs of tape machine frequency responses, we can mimic them using EQ. There are plenty of graphs at endino.com/graphs – we base ours on the Ampex MM-1200 2-inch 16-track, using a high-pass at 46Hz with a Q of 0.90, and an AnaPeak band at 140Hz, -2.0dB, 2.4Q.

4 > Now copy both the Satson CM and the EQ to the Bass channel, to give it the same flavour. We don't need Satson CM to be quite as aggressive on the bass, so set High Pass to 1 and Low Pass to 9. The bass track sounds good, but it's focused more on the upper midrange than it is on serious bass frequencies, so let's tackle that next...

5 > Add another IIEQ Pro CM on the bass track. This time we'll recreate the famous low-end trick associated with vintage Pultec EQ units. Set Bands 1 and 2 to Low Shelf, both at 200Hz, then boost with 1 and cut with 2 by 10dB. The result is... a flat EQ curve that does nothing to the sound! Kinda pointless, right? Now hit the Series button to switch to Parallel mode.

6 > The bands create a sub bass boost *and* a lower mid cut, giving huge perceived bass and warmth. You can adjust settings separately to hone the effect. Copy Satson CM and the tape-curve IIEQ Pro CM from the Bass track to the remaining tracks (including the master), then finalise with your favourite analogue-style EQ, tape sim and compression on the master bus.

## Five classic analogue-style EQs

**SoundToys Sie-Q**
**$99**
Clearly not content to slap their name on yet another 1073 or Pultec clone, Soundtoys' first EQ plugin recreates the W295b EQ module from a 60s Siemens Sitral console. Improvements include fully sweepable gain rather than stepped selections, and adjustable drive for adding juicy analogue saturation.
**soundtoys.com**

**Slate Digital Virtual Mix Rack**
**cm214 » 9/10 » £129**
No roster of analogue EQs would be complete without an SSL-alike, and FG-S is one of the finest, found in Slate Digital's Virtual Mix Rack package. It works just like the real thing, with a high-pass filter, low and high shelves switchable to bell shapes, and two midrange peaking filters. Powerful, clean and musical.
**slatedigital.com**

**PSP Audioware ClassicQ**
**$69**
Developers might be fixated on recreating gear from big-name brands like SSL, API and Neve emulations, but Polish DSP veterans PSP picked out a British Amek EQ as the inspiration for ClassicQ. It features saturation and a switchable transformer simulation for subtle-sounding low-end bloom.
**pspaudioware.com**

**Sonimus Burnley 73**
**cm226 » 10/10 » $59**
The Neve 1073 is one of the all-time greats, comprising a preamp with incredible-sounding drive and powerful, muscular EQ. With a spot-on sound and smart modern additions like band bypass, pop-up parameter labelling, and level-compensated drive, this recreation from Sonimus is hard not to recommend.
**sonimus.com**

**Sly-Fi Axis**
**$99**
An off-shoot of the main Kush brand, Sly-Fi's 'triple-mode' EQ is inspired by the sound of two classic API hardware equalisers, switchable with the A/B setting. A third mode, UBK, does away with the stepped frequency and gain selectors, allowing completely smooth adjustment. The star of the show, though, is the sumptuous saturation.
**slyfidigital.com**

## > Step by step

## 8. Sending sounds back in time for the ultimate analogue journey

1 > Let's go all-out and send a pristinely recorded production more than a century back in time. Music production was in its infancy, and recordings were made acoustically, channelling sound waves down a funnel or tube to vibrate a mechanical cutter pressed on the rotating cylinder or disc. Playback used the same principle, but in reverse – there were no microphones, mixers, amps, or speakers!

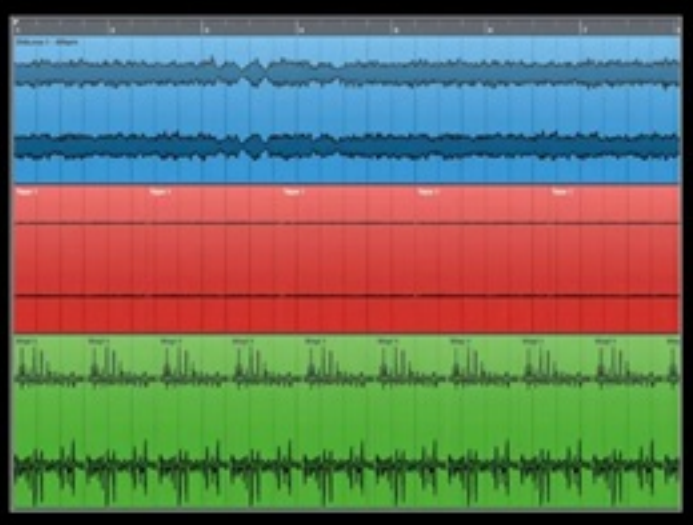
2 > To get started, load OldLoop 1.wav in an 85bpm project and loop it – we're going to use free **cm** Plugins to recreate those turn of the 20th-century vibes. First things first: put a monoising plugin (such as DeeMonitor with Mid at 0%) on the master bus, as the first recordings were strictly mono.

3 > Bass and treble were practically non-existent in antique recordings. To mimic this, place a Philta CM next on the master bus with Lowpass at 3000Hz and Highpass at 300Hz. Set both Slopes to 48dB for a brutally sharp cutoff. You may need to fine-tune the exact filter settings to suit the sound source, but this is a good starting point.

4 > Next, we'll recreate the poor dynamic range and distortion performance of wax and shellac mediums. Add Premix CM before Philta CM (placing it after would reintroduce frequencies that the filter has shaved off), and increase Gain to 3 o'clock, setting Output to 9 o'clock to compensate. To give the loop more presence, you can boost the Mid and High knobs to taste.

5 > Mechanical audio playback is prone to wow and flutter – variations in playback speed affecting timing and pitch. We can recreate this with a modulated delay line. Add PSP Audioware cmDelay before PreMix CM. Click its Dry label to disable the dry part of the sound, and click the Filters label to turn off the filters. Set the Delay Time to 5ms and FB to 0.

6 > Now turn LFO Depth up to maximum, and dial in the LFO Rate to set up a slow wow or a rapid flutter as you prefer – we set it to the maximum value of 15 for flutter. Pull LFO Depth down when you're done – 75 is enough for us. If you want wow *and* flutter, just use two cmDelays in series, with different settings.



7 > What retro recreation would be complete without noise, clicks and pops? Put Vinyl 1.wav and Tape 1.wav on new tracks, disable any tempo-synced looping/warping options, then loop or duplicate the clips. The idea is that the loops do *not* restart with each bar, as real mechanical noise would be unpredictable. Adjust the tape and noise levels to taste.

8 > We use sidechain compression to make OldLoop1 duck with each of Vinyl 1's pops and clicks, and to make Tape 1 duck in response to OldLoop1. Have fun running other sounds through this processing chain and trying different tape/noise loops. Don't forget to dial in plugin parameters such as filter cutoff frequencies to suit your sound source.

POWER TIP

### >And in reverse order...

You may have noticed that we created most of our retro processing chain in reverse order, from end to beginning. This is because the processing is so extreme that we need to hear how each stage will be affected by the next in the chain as we're dialling it in. For example, you might select a noise loop full of sizzling trebly hiss and bassy hum that sounds great on its own, but once you apply aggressive top-and-tail filtering, those elements will be all but silenced. Therefore, you'd need to audition a few loops with the filter in place to find ones that cut through the processing.

# Saturation in the mix

Saturation is a defining factor in 'that analogue sound'. It can make sounds fat and beefy, smooth and warm, or crunchy and edgy. Placed strategically on single tracks and buses, it can produce a more cohesive mix.

The most obvious effect is the addition of harmonic content, most easily heard on sounds that don't have much to begin with, such as subby sine basses. By adding upper harmonics, a deep bass that vanishes on smaller speaker systems magically becomes audible. Generally, transistor distortion adds sharpness and crunch, and valve and tape are associated with warmth.

Sharp transients can be bothersome, but saturation can effectively control them. Used gently, it can invisibly reduce peaks, gaining headroom, while stronger settings audibly soften and 'smoosh' transients while retaining – even emphasising – lower punch and impact. Try it on ticky hi-hats for an audible example.

**It's often best to add saturation early in the signal chain**

Saturation is also a form of dynamic control, attenuating loud parts of the signal. Tape in particular has squashy, compression-like qualities, as do some valve saturations.

Each saturation device affects harmonics, transients and dynamics differently, and getting to know yours will help you pick the right one. For example, transformers saturate more at low frequencies. Since transformers are found in the input and output stages of much classic gear, the accumulation of distortion can significantly warm your bottom end.

Where you place saturation can make a profound difference. There are no concrete rules, but it generally works well at the start of the chain, perhaps after corrective EQ. For heavier saturation, placing compression first gives more consistent distortion. Saturation placed on group and master buses can add cohesion, but beware that more complex input signals result in increased intermodulation distortion, which is pretty unmusical. This is why single notes distort nicely while chords generate mush and noise.

## > Step by step 9. Stacking saturation for a punchier mixdown

**1** > By using multiple stages of gentle saturation throughout a mix, you can add firmness, density and texture. Start by loading the WAVs 'HipHop...' onto audio tracks in a 95bpm project. Let's start with the bass – solo it and you'll hear that it's basically a sine wave. There's an extra four bars on the bass track containing one sustained note, which we can use as a test signal for frequency analysis.

**2** > On the bass track, add an analyser plugin like Vengeance's Scope CM in Spectrum mode to see the single harmonic of the bassline. Add 6X-500 CM before it and increase Gain as shown. This preamp transistor distortion adds odd harmonics. Now bypass it and do the same with Premix CM – we can see that this valve sim adds plenty of even harmonics, too.

**3** > Next we try Slate Digital's Virtual Tape Machines, which adds a strong third harmonic, along with tape artefacts like hiss. This one works best for us, with Input set to +15dB. Even this fairly mild saturation benefits our bassline, as the added harmonics make it audible on sound systems that can't fully reproduce the deep fundamental tone.

**4** > Now solo the snare track and add Soundtoys Decapitator to it. Despite the brutal name, it gives great preamp-style saturation if you don't overdo it. Set Drive to 5 and click through the five distortion Styles (A, E, N, T and P) to hear their differences – hear how they respond dynamically, too. We like mode T, a triode valve sim – pull Drive back to 2.5 for a more subtle effect.

**5** > The distortion has had two undesirable effects: the snare is louder and its transient not as punchy. To fix it, pull Output down to -4dB to compensate for the level change, and pull down Mix to 85% (click the knob labels to show their values) to blend some of the dry, undistorted signal back in. This parallel distortion technique is very useful for preserving punch.

**6** > On the kick drum, we use Decapitator once again, with Drive at 3, 85% Dry/Wet, and Style A, modelling an Ampex 350 tape preamp. Next, route tracks 1-5 to a group bus so we can glue them together with more saturation with Slate Digital's Virtual Tape Machines, with Input at 9dB. For a final touch, we use a tape emulation on the master bus. The end result is a deeper, fatter, more analogue mix.

# Three of the best vinyl and noise plugins

**AudioThing Vinyl Strip**
**cm226 » 9/10 » $59**
This well-equipped channel strip is a must-have tool for injecting old-school flavour and vibe into your channels. Of Vinyl Strip's six reorderable modules (which include drive, compression, sampler crunch, tone tilting and reverb), our favourite is the fully stocked Vinyl section, for mixing in customisable vinyl noise and crackle into the mix.
**audiothing.net**

**Vengeance-Sound VPS Vinylizer**
**cm214 » 9/10 » £85 (bundle)**
Part of the Vengeance-Sound's VPS Essential FX Bundle 2, this virtual 'ageing' toolkit places a plethora of noise-outputting generators at your fingertips. Dial in noise hum, rumble, crackle and scratches; customise tone and density; and even change the speed of your virtual record, and switch between line/phono inputs.
**vengeance-sound.com**

**iZotope Vinyl**
**FREE**
Throw this fantastic freebie on a track to place it onto a virtual record deck, complete with all the artefacts associated with the medium. You can crank up your warp speed, hum, dust, wear, mechanical noise and age, meaning that you don't have to use it just for simulating the quirks of the vinyl medium. Vintage vibes incoming!
**izotope.com**

## > Step by step 10. Essential noise layering

1 > Back in the day, producers were desperate to remove the ever-present noise outputted by analogue equipment, but nowadays the lack of it can expose a sound as overly pristine and 'digital'. The solution? Blend analogue hiss, hum or noise signals (or all of the above) alongside your shiniest sounds to fake an analogue aesthetic! We'll start with this digital-sounding track.

2 > This technique isn't rocket science: drop a sample of noise onto an audio track, load the noise sample into a sampler for MIDI playback, or insert a dedicated noise-outputting plugin on a track (see to the right for three of our faves). Use the track's volume fader to mix the noise in to taste – find the point where it's not too overpowering in the mix, but is missed when muted.

3 > A consistent stream of noise can sound jarring, so help your noise interact with drums and rhythmic material. In this example, the peaks of the main drum beat are triggering subtle sidechain compression over the noise, giving it gentle movement. For cohesion, route the channel into your drum bus, so it gets processed with the rest of the beats.

4 > There's just *something* about the warm, dusty sound of a sample ripped from vinyl! To make this clap sound like it's been lifted from a record, we use the clap's MIDI note to trigger a vinyl crackle sample layer. The rhythms within pops and clicks can add flavour to a groove, so move the sample start point around until you find the perfect snippet of skip.

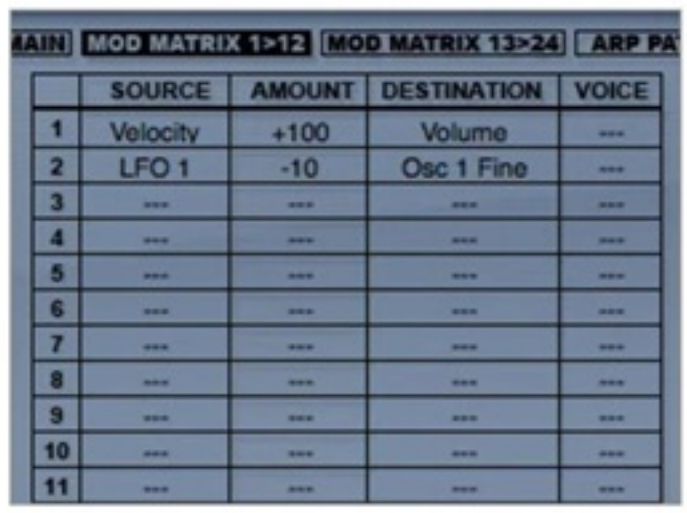
5 > As we've mentioned, different types of noise used to build up and compound due to the combined analogue signals and circuits in an all-hardware studio. So, for a truly analogue-sounding mix, pile up different types of noise signals on various channels. We've added a hum signal into the mix using iZotope Vinyl.

6 > Here's a riff from our Bazille CM synth, and an analogue hiss signal from AudioThing VinylStrip. A gate plugin is loaded on the hiss channel, and the riff signal is fed into the gate's sidechain input. The gate quickly opens over riff transients, adding tiny bursts of noise at the start of each note, 'ageing' the modern sound.

> Step by step

## 11. Creating pitch-wavering analogue chords

1 > Let's reverse-engineer the imperfections of an old analogue synth and create warm, retro chords using **cm** Plugins and a little know-how. Call up Synapse Audio's Dune CM on a new MIDI track in your DAW, hit its Bank B button to initialise the synth, and drop Retro Chords.mid onto this MIDI channel to trigger the synth. Plonk a limiter on the master bus to prevent clipping.




2 > The oscillator section's Osc Mix knob is fully anticlockwise by default, giving us a simple single-oscillator sawtooth. First, we'll use an LFO to slowly modulate the pitch of this oscillator, to emulate the unstable pitch drift and inconsistency of an analogue oscillator. Head into the Mod Matrix 1>12 tab, then set row 2's Source to LFO 1, an Amount of -10, and a Destination of Osc 1 Fine.

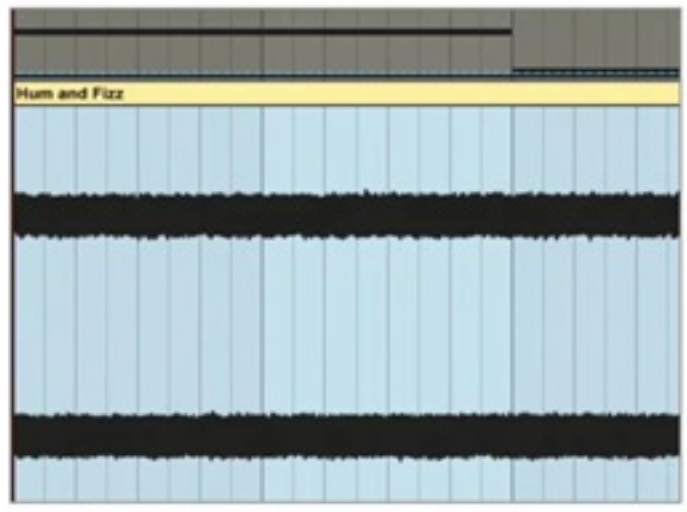
3 > This pitch modulation is way too fast, so head over to the LFO section and set LFO 1's Rate to the minimum 0.1Hz. This LFO is currently set to a Sine shape, which is a little formulaic – switch the shape to the bottom random type for more unpredictable pitch wavering. Next, turn the Noise oscillator's Level up to a subtle 6% for slight top-end sizzle.

4 > Our patch's pitch now has a subtle 'beating' character, yet its timbre is still rather bright and digital-sounding – but instead of reaching for an obvious EQ or filter, let's use the fantastic PSP cmDelay (again, free in **cm** Plugins) to add a bit of darkness, pitch modulation and analogue warmth. Load up a new instance on the Dune CM channel.

5 > As we only want to hear the wet signal, click on the Output » Dry header to deactivate the dry signal. Now we'll shape the chords' tone with cmDelay's high- and low-pass filters – by default, these are band-passing our sound a little too much, so dial back the low-pass knob to 20Hz and high-pass to 10kHz. Push Input Gain up to 4dB to level the output against the unprocessed signal.

6 > The tape delay-emulating cmDelay features an LFO for modulating the Delay Time, which in turn modulates the pitch of the delays – an effect we can use to add further subtle pitch 'warbling' to our synth chords. Crank up the LFO Depth knob to 80, then set a fast Delay Time of around 0004. Now alter the LFO Rate to taste – we like a setting of between 0.5Hz and 1Hz. Very Boards of Canada!

7 > For ambience, load a fresh cmDelay on a new aux return track, then send the Dune CM channel to the return by 0dB. Deactivate cmDelay's Dry signal, then set a 130ms Delay Time and Feedback (FB) of around 88. Pan the main synth channel slightly out to 6L, then pan the delay return channel to 35R. Voila – simple, retro-style width without any glossy widening techniques!




8 > As we've explored elsewhere, hum and noise layers are great for adding the illusion of real analogue flavour within a mix. Drop Hum and Fizz.wav – a combined audio bounce of system hum and an analogue noise floor – onto a new audio channel. Route the chords and noise to a new group channel, remove the chords' delay send, then pipe the entire group to the delay aux by 0dB.

9 > For some gentle gluing saturation, insert HoRNet's Graffio CM on the group, and set Saturator Gain to 8dB and output Level to -3dB. To finish, we'll load Satson CM after Graffio CM before pulling its Low Pass knob back to 3 o'clock, removing any trace of treble and digital harshness. To finish, A/B between the init Dune CM patch and the final sound to hear the obvious ageing! **cm**

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